Cisco Unified IP Phone 7906G and 7911G Administration Guide for Cisco Unified Communications Manager 8.0 (SCCP and SIP)

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Preface

Overview

Cisco Unified IP Phone 7906G and 7911G Administration Guide for Cisco Unified Communications Manager 8.0 provides the information you need to understand, install, configure, manage, and troubleshoot the Cisco Unified IP Phones 7906G and 7911G in a Voice-over-IP (VoIP) network.

Because of the complexity of a Unified Communications network, this guide does not provide complete and detailed information for procedures that you need to perform in Cisco Unified Communications Manager (formerly Cisco Unified CallManager) or other network devices. See the Related Documentation, page 2 for a list of related documentation.

Audience

Network engineers, system administrators, or telecom engineers should review this guide to learn the steps required to properly set up the Cisco Unified IP Phones 7906G and 7911G on the network.

The tasks described are administration-level tasks and are not intended for end-users of the phones. Many of the tasks involve configuring network settings and affect the phone's ability to function in the network.

Because of the close interaction between the Cisco Unified IP Phone and Cisco Unified Communications Manager, many of the tasks in this manual require familiarity with Cisco Unified Communications Manager.

Organization

This manual is organized as follows:

Chapter 1, An Overview of the Cisco Unified IP Phone	Provides a conceptual overview and description of the Cisco Unified IP Phone.
Chapter 2, Preparing to Install the Cisco Unified IP Phone on Your Network	Describes how the Cisco Unified IP Phone interacts with other key Unified Communications components, and provides an overview of the tasks required prior to installation.
Chapter 3, Setting Up the Cisco Unified IP Phone	Describes how to properly and safely install and configure the Cisco Unified IP Phone on your network.

Chapter 4, Configuring Settings on the Cisco Unified IP Phone	Describes how to configure network settings, verify status, and make global changes to the Cisco Unified IP Phone.
Chapter 5, Configuring Features, Templates, Services, and Users	Provides an overview of procedures for configuring telephony features, configuring directories, configuring phone button and softkey templates, setting up services, and adding users to Cisco Unified Communications Manager.
Chapter 6, Customizing the Cisco Unified IP Phone	Explains how to customize phone ring sounds, background images, and the phone idle display at your site.
Chapter 7, Viewing Model Information, Status, and Statistics on the Cisco Unified IP Phone	Explains how to view model information, status messages, network statistics, and firmware information from the Cisco Unified IP Phone.
Chapter 8, Monitoring the Cisco Unified IP Phone Remotely	Explains how to obtain status information about the phone using the phone's web page.
Chapter 9, Troubleshooting and Maintenance	Provides tips for troubleshooting the Cisco Unified IP Phone.
Appendix A, Providing Information to Users	Provides suggestions for setting up a website for providing users with important information about their Cisco Unified IP Phones.
Appendix B, Feature Support by Protocol for Cisco Unified IP Phone 7906G and 7911G	Provides information about feature support for the Cisco Unified IP Phone using the SCCP or SIP protocol.
Appendix C, Supporting International Users	Provides information about setting up phones in non-English environments.
Appendix D, Technical Specifications	Provides technical specifications of the Cisco Unified IP Phone.
Appendix E, Adding a User to Cisco Unified Communications Manager	Provides procedures for basic administration tasks such as adding a user and phone to Cisco Unified Communications Manager and then associating the user to the phone.

Related Documentation

For more information about Cisco Unified IP Phones or Cisco Unified Communications Manager, refer to the following publications:

Cisco Unified IP Phones 7906G and 7911G

These publications are available at the following URL:

http://www.cisco.com/en/US/products/hw/phones/ps379/tsd_products_support_series_home.html

- *Cisco Unified IP Phone 7906G Installation Guide*
- *Cisco Unified IP Phone 7911G Installation Guide*
- *Cisco Unified IP Phone 7906G and 7911G Phone Guide*
- *Cisco Unified IP Phone Features A–Z*
- *Regulatory Compliance and Safety Information for Cisco Unified IP Phones*
- *Installing the Universal Wall Mount Kit for the Cisco Unified IP Phone*
- *Open Source License Notices for the Cisco Unified IP Phones 7900 Series*

Cisco Unified Communications Manager

Related publications are available at the following URL:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/tsd_products_support_series_home.html

Cisco Unified Communications Manager Business Edition

Related publications are available at the following URL:

http://www.cisco.com/en/US/products/ps7273/tsd_products_support_series_home.html

Obtaining Documentation, Obtaining Support, and Security Guidelines

For information about obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and recommended aliases and general Cisco documents, see the monthly What's New in Cisco Product Documentation, which also lists all new and revised Cisco technical documentation, at:

http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html

Cisco Product Security Overview

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

Further information regarding U.S. export regulations may be found at: http://www.access.gpo.gov/bis/ear/ear_data.html.

Document Conventions

This document uses the following conventions:

Convention	Description
boldface font	Commands and keywords are in **boldface**.
italic font	Arguments for which you supply values are in *italics*.
[ ]	Elements in square brackets are optional.
{ x | y | z }	Alternative keywords are grouped in braces and separated by vertical bars.
[x | y | z]	Optional alternative keywords are grouped in brackets and separated by vertical bars.
string	A nonquoted set of characters. Do not use quotation marks around the string or the string will include the quotation marks.
`screen` font	Terminal sessions and information the system displays are in `screen` font.

Convention	Description
boldface screen font	Information you must enter is in **boldface screen** font.
italic screen font	Arguments for which you supply values are in *italic screen* font.
^	The symbol ^ represents the key labeled Control—for example, the key combination ^D in a screen display means hold down the Control key while you press the D key.
< >	Nonprinting characters, such as passwords are in angle brackets.

Note Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the publication.

Caution Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

Warning **Means danger. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.**

CHAPTER 1

An Overview of the Cisco Unified IP Phone

The Cisco Unified IP Phone 7906G and 7911G provide voice communication over an Internet Protocol (IP) network. It functions much like a standard digital business telephone, allowing you to place and receive phone calls and to access features such as mute, hold, transfer, and speed dial. In addition, because the phone is connected to your data network, it offers enhanced productivity features, including access to network information, XML applications, and customizeable features.

A Cisco Unified IP Phone, like other network devices, must be configured and managed. These phones encode G.711a, G.711u, G.722, G.729a, G.729ab, iLBC, and decode G.711a, G.711u, G.722, iLBC, G.729, G.729a, G.729b, and G.729ab. These phones also support uncompressed wideband (16 bits, 16 kHz) audio.

This chapter includes the following topics:

- Understanding the Cisco Unified IP Phones 7906G and 7911G, page 1-1
- What Networking Protocols are Used?, page 1-3
- IPv6 Support on Cisco Unified IP Phones, page 1-7
- What Features are Supported?, page 1-8
- Understanding Security Features for Cisco Unified IP Phones, page 1-10
- Overview of Configuring and Installing Cisco Unified IP Phones, page 1-18

Caution Using a cell, mobile, or GSM phone, or two-way radio in close proximity to a Cisco Unified IP Phone might cause interference. For more information, refer to the manufacturer documentation of the interfering device.

Understanding the Cisco Unified IP Phones 7906G and 7911G

The Cisco Unified IP Phones 7906G and 7911G are basic IP phones designed for cubicles, classrooms, factory floors, warehouses, lobbies, and any other location where the phone either complements the user's set of communication devices or is seldom used. The Cisco Unified IP Phones 7906G and 7911G:

- Provide a graphical display with dynamic softkeys, icons, and scrollable directories for easy access to a core set of business features
- Support up to six calls on one directory number
- Support inline power for both Cisco inline power or IEEE 802.3af Power over Ethernet
- Support enhanced security features including:

  - Manufacturing and field installable certificates
  - Secure Media and Signaling
  - Authenticated Configuration
- Support enhanced calling features plus audio and text XML applications
- Include an integrated 10/100 Mbit Ethernet switch for connecting a PC, thereby preserving the advantage of one cable pull per location (applies to Cisco Unified IP Phone 7911G only)

Figure 1-1 shows the main components of the Cisco Unified IP Phones 7906G and 7911G.

Figure 1-1 Cisco Unified IP Phones 7906G and 7911G

Table 1-1 describes the buttons on the Cisco IP Phones 7906G and 7911G.

Table 1-1 Features on the Cisco Unified IP Phone 7906G and 7911G

1	Phone screen	Displays phone features such as phone number, call status, and softkeys.
2	Cisco Unified IP Phone series	Indicates your Cisco Unified IP Phone model series.
3	Softkeys	Each softkey activates a softkey option displayed on your phone screen.
4	Navigation button	Allows you to scroll through menu items and highlight items. When the phone is on-hook, displays your Speed Dials.
5	Applications menu button	Displays the Applications menu that provides access to a voice messaging system, phone logs and directories, settings, and services.
6	Hold button	Places the active call on hold, resumes a call on hold, and switches between an active call and a call on hold.
7	Keypad	Allows you to dial phone numbers, enter letters, and choose menu items.
8	Volume button	Controls the handset, headset, speaker, and ringer volume.
9	Handset	Functions like a traditional handset. The light strip at the top of the handset blinks when the phone rings and stays lit if there is a new voice message (depending on your voice messaging system).
10	Footstand	Allows the phone to stand at a convenient angle on a desk or table. Also may be removed for wall mounting to mounting screws or to a Cisco Unified IP Phone wall mount kit.

What Networking Protocols are Used?

Cisco Unified IP Phones support several industry-standard and Cisco networking protocols required for voice communication. Table 1-2 provides an overview of the supported networking protocols on the Cisco Unified IP Phones 7906G and 7911G.

Table 1-2 Supported Networking Protocols on the Cisco Unified IP Phone

Networking Protocol	Purpose	Usage Notes
Bootstrap Protocol (BootP)	BootP enables a network device such as the Cisco Unified IP Phone to discover certain startup information, such as its IP address.	If you are using BootP to assign IP addresses to the Cisco Unified IP Phone, the BOOTP Server option shows "Yes" in the network configuration settings on the phone.
Cisco Discovery Protocol (CDP)	CDP is a device-discovery protocol that runs on all Cisco-manufactured equipment. Using CDP, a device can advertise its existence to other devices and receive information about other devices in the network.	The Cisco Unified IP Phone uses CDP to communicate information such as auxiliary VLAN ID, per port power management details, and Quality of Service (QoS) configuration information with the Cisco Catalyst switch.
Cisco Peer-to-Peer Distribution Protocol (CPPDP)	CPPDP is a Cisco proprietary protocol used to form a peer-to-peer hierarchy of devices. CPPDP is also used to copy firmware or other files from peer devices to neighboring devices.	CPPDP is used by the Peer Firmware Sharing feature.
Dynamic Host Configuration Protocol (DHCP)	DHCP dynamically allocates and assigns an IP address to network devices. DHCP enables you to connect an IP phone into the network and have the phone become operational without your needing to manually assign an IP address or to configure additional network parameters.	DHCP is enabled by default. If disabled, you must manually configure the IP address, subnet mask, gateway, and a TFTP server on each phone locally. Cisco recommends that you use DHCP custom option 150. With this method, you configure the TFTP server IP address as the option value. For additional information about DHCP configurations, refer to the Dynamic Host Configuration Protocol and Cisco TFTP in the *Cisco Unified Communications Manager System Guide*.
HyperText Transfer Protocol (HTTP)	HTTP is the standard way of transferring information and moving documents across the Internet and the World Wide Web.	The Cisco Unified IP Phones use HTTP for the XML services and for troubleshooting purposes. Cisco Unified IP Phones do not support the use of IPv6 addresses in the URL. You cannot use literal IPv6 addresses in the URL or hostname which maps to a IPv6 address in the directories and services button.
Hypertext Transfer Protocol Secure (HTTPS)	Hypertext Transfer Protocol Secure (HTTPS) is a combination of the Hypertext Transfer Protocol with the SSL/TLS protocol to provide encryption and secure identification of servers.	Web applications with both HTTP and HTTPS support have two URLs configured. Cisco Unified IP Phones that support HTTPS choose the HTTPS URL out of the two URLs.

Table 1-2 Supported Networking Protocols on the Cisco Unified IP Phone (continued)

Networking Protocol	Purpose	Usage Notes
IEEE 802.1X	The IEEE 802.1X standard defines a client-server-based access control and authentication protocol that restricts unauthorized clients from connecting to a LAN through publicly accessible ports. Until the client is authenticated, 802.1X access control allows only Extensible Authentication Protocol over LAN (EAPOL) traffic through the port to which the client is connected. After authentication is successful, normal traffic can pass through the port.	The Cisco Unified IP Phone implements the IEEE 802.1X standard by providing support for the following authentication methods: EAP-FAST, EAP-TLS, and EAP-MD5. When 802.1X authentication is enabled on the phone, you should disable the PC port and voice VLAN. See Supporting 802.1X Authentication on Cisco Unified IP Phones, page 1-16 for additional information.
Internet Protocol (IP)	IP is a messaging protocol that addresses and sends packets across the network.	To communicate using IP, network devices must have an assigned IP address, subnet, and gateway. IP addresses, subnets, and gateways identifications are automatically assigned if you are using the Cisco Unified IP Phone with Dynamic Host Configuration Protocol (DHCP). If you are not using DHCP, you must manually assign these properties to each phone locally. The Cisco Unified IP Phone supports concurrent IPv4 and IPv6 addresses. Configure the IP addressing mode (IPv4 only, IPv6 only, and both IPv4 and IPv6) in Cisco Unified Communications Manager Administration. For more information, refer to Internet Protocol Version 6 (IPv6) in the *Cisco Unified Communications Manager Features and Services Guide*.
Link Layer Discovery Protocol (LLDP)	LLDP is a standardized network discovery protocol (similar to CDP) that is supported on some Cisco and third-party devices.	The Cisco Unified IP Phone supports LLDP on the PC port.
Link Layer Discovery Protocol-Media Endpoint Devices (LLDP-MED)	LLDP-MED is an extension of the LLDP standard developed for voice products.	The Cisco Unified IP Phone supports LLDP-MED on the SW port to communicate information such as: • Voice VLAN configuration • Device discovery • Power management • Inventory management For more information about LLDP-MED support, see the *LLDP-MED and Cisco Discovery Protocol* white paper: http://www.cisco.com/en/US/tech/tk652/tk701/technologies_white_paper0900aecd804cd46d.shtml

Table 1-2 Supported Networking Protocols on the Cisco Unified IP Phone (continued)

Networking Protocol	Purpose	Usage Notes
Real-Time Transport Protocol (RTP)	RTP is a standard protocol for transporting real-time data, such as interactive voice and video, over data networks.	Cisco Unified IP Phones use the RTP protocol to send and receive real-time voice traffic from other phones and gateways.
Real-Time Control Protocol (RTCP	RTCP works with RTP to provide QoS data (such as jitter, latency, and round trip delay) on RTP streams.	RTCP is disabled by default, but you can enable it on a per phone basis by using Cisco Unified Communications Manager. For more information, see Network Configuration, page 4-28.
Secure Real-Time Transport Protocol (SRTP)	SRTP is available in addition to RTP. SRTP adds security by encrypting media streams during data transport.	For SRTP to work, the phone or phones being called must also support SRTP or else those phones cannot decrypt the secure media stream.
Session Initiation Protocol (SIP)	SIP is the Internet Engineering Task Force (IETF) standard for multimedia conferencing over IP. SIP is an ASCII-based application-layer control protocol (defined in RFC 3261) that can be used to establish, maintain, and terminate calls between two or more endpoints.	Like other VoIP protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. *Signaling* allows call information to be carried across network boundaries. *Session management* provides the ability to control the attributes of an end-to-end call. You can configure the Cisco Unified IP Phone to use either SIP or Skinny Client Control Protocol (SCCP). Cisco Unified IP Phones do not support the SIP protocol when the phones are operating in IPv6 address mode.
Skinny Client Control Protocol (SCCP)	SCCP includes a messaging set that allows communications between call control servers and endpoint clients such as IP Phones. SCCP is proprietary to Cisco Systems.	Cisco Unified IP Phones use SCCP for call control. You can configure the Cisco Unified IP Phone to use either SCCP or Session Initiation Protocol (SIP).
Session Description Protocol (SDP)	SDP is the portion of the SIP protocol that determines which parameters are available during a connection between two endpoints. Conferences are established by using only the SDP capabilities that are supported by all endpoints in the conference.	SDP capabilities, such as codec types, DTMF detection, and comfort noise, are normally configured on a global basis by Cisco Unified Communications Manager or Media Gateway in operation. Some SIP endpoints may allow these parameters to be configured on the endpoint itself.
Transmission Control Protocol (TCP)	TCP is a connection-oriented transport protocol.	Cisco Unified IP Phones use TCP to connect to Cisco Unified Communications Manager and to access XML services.
Transport Layer Security (TLS)	TLS is a standard protocol for securing and authenticating communications.	When security is implemented, Cisco Unified IP Phones use the TLS protocol when securely registering with Cisco Unified Communications Manager. For more information, refer to the *Cisco Unified Communications Manager Security Guide.*

Table 1-2 Supported Networking Protocols on the Cisco Unified IP Phone (continued)

Networking Protocol	Purpose	Usage Notes
Trivial File Transfer Protocol (TFTP)	TFTP allows you to transfer files over the network. On the Cisco Unified IP Phone, TFTP enables you to obtain a configuration file specific to the phone type.	TFTP requires a TFTP server in your network, which can be automatically identified from the DHCP server. If more than one TFTP server is running in your network, you must manually assign a TFTP server to each phone locally. For more information, refer to Cisco TFTP in the *Cisco Unified Communications Manager System Guide*.
User Datagram Protocol (UDP)	UDP is a connectionless messaging protocol for delivery of data packets.	Cisco Unified IP Phones receive and process UDP messages.

Related Topics

- Understanding Interactions with Other Cisco Unified Communications Products, page 2-1
- Understanding the Phone Startup Process, page 2-5
- Network Configuration Menu, page 4-5

IPv6 Support on Cisco Unified IP Phones

The Cisco Unified IP Phone uses the internet protocol to provide voice communication over the network. Previous to Cisco Unified Communications Manager release 7.1, only the internet protocol version 4 (IPv4) was supported. However because IPv4 uses a 32-bit address, it cannot meet the increased demands for unique IP addresses for all devices that can connect to the internet. Therefore, Internet Protocol version 6 (IPv6), an updated version of the current Internet Protocol, IPv4 is now supported. IPv6 uses a 128-bit address and provides end-to-end security capabilities, enhanced Quality of Service (QoS), and increased number of available IP addresses.

The Cisco Unified IP Phone supports IPv4 only addressing mode, IPv6 only addressing mode, as well as an IPv4/IPv6 dual stack addressing mode. In IPv4, you can enter each octet of the IP address on the phone in dotted decimal notation; for example, 192.240.22.5. In IPv6, you can enter each octet of the IP address in hexadecimal notation with each octet separated by a colon; for example, 2005:db8:0:1:ef8:9876:ba72:dc9a. The phone truncates and removes leading zeros when it displays the IPv6 address.

Cisco Unified IP Phones support both IPv4 and an IPv6 address transparently, so users can handle all calls on the phone to which they are accustomed. Cisco Unified IP Phones support the use of IPv6 only with Cisco Unified Communications Manager release 7.1 and only with the Skinny Call Control Protocol (SCCP).

Cisco Unified IP Phones do not support URLs with IPv6 addresses in the URL. This affects all IP Phone Service URLs, including services, directories, messages, help, and any restricted web services that require the phone to use the HTTP protocol to validate the credentials with the Authentication URL. If you configure Cisco Unified IP Phone services for Cisco Unified IP Phones, you must configure the phone and the servers that support the phone service with IPv4 addresses.

If you configure IPv6 Only as the IP Addressing Mode for phones that are running SIP, the Cisco TFTP service overrides the IP Addressing Mode configuration and uses IPv4 Only in the configuration file.

For more information on deploying IPv6 in your Cisco Unified Communications network, refer to Internet Protocol Version 6 (IPv6) in *Cisco Unified Communications Manager Features and Services Guide* and *Deploying IPv6 in Unified Communications Networks with Cisco Unified Communications Manager.*

What Features are Supported?

Cisco Unified IP Phones 7906G and 7911G function much like traditional analog phones, allowing you to place and receive phone calls. In addition to traditional telephony features, each Cisco Unified IP Phone includes features that enable you to administer and monitor the phone as a network device.

This section includes the following topics:

- Feature Overview, page 1-8
- Configuring Telephony Features, page 1-9
- Configuring Network Parameters Using the Cisco Unified IP Phone, page 1-9
- Providing Users with Feature Information, page 1-9

Feature Overview

Cisco Unified IP Phones provide core business features, such as call forwarding and transferring, redialing, speed dialing, conference calling, and voice messaging system access. Cisco Unified IP phones also provide a variety of other features. For an overview of the telephony features that the Cisco Unified IP Phone supports, see Telephony Features Available for the Cisco Unified IP Phone, page 5-1.

As with other network devices, you must configure Cisco Unified IP Phones to prepare them to access Cisco Unified Communications Manager and the rest of the IP network. Using DHCP, you have fewer settings to configure on a phone, but if your network requires it, you can manually configure an IP address, TFTP server, subnet information, etc. For instructions on configuring the network settings on the Cisco Unified IP Phones, see Configuring Settings on the Cisco Unified IP Phone, page 4-1.

Cisco Unified IP Phones can interact with other services and devices on your IP network to provide enhanced functionality. For example, you can integrate Cisco Unified Communications Manager with the corporate Lightweight Directory Access Protocol 3 (LDAP3) standard directory to enable users to search for co-workers contact information directly from their IP phones. Or, you can also use XML to enable users to access information such as weather, stocks, quote of the day, and other web-based information. For information about configuring such services, see Configuring Corporate and Personal Directories, page 5-16 and Setting Up Services, page 5-20.

Finally, because the Cisco Unified IP Phone is a network device, you can obtain detailed status information from it directly. This information can assist you with troubleshooting any problems users might encounter when using their IP phones. See Viewing Model Information, Status, and Statistics on the Cisco Unified IP Phone, page 7-1, for more information.

Related Topics

- Configuring Settings on the Cisco Unified IP Phone, page 4-1
- Configuring Features, Templates, Services, and Users, page 5-1
- Troubleshooting and Maintenance, page 9-1

Configuring Telephony Features

You can modify certain settings for the Cisco Unified IP Phone from the Cisco Unified Communications Manager Administration application. Use this web-based application to set up phone registration criteria and calling search spaces, to configure corporate directories and services, and to modify phone button templates, among other tasks. See Telephony Features Available for the Cisco Unified IP Phone, page 5-1 and *Cisco Unified Communications Manager Administration Guide* for additional information.

For more information about the Cisco Unified Communications Manager Administration application, refer to Cisco Unified Communications Manager documentation, including *Cisco Unified Communications Manager System Guide*. You can also use the context-sensitive help available within the application for guidance.

You can access Cisco Unified Communications Manager documentation at this location:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/tsd_products_support_series_home.html

You can access Cisco Unified Communications Manager Business Edition documentation at this location:

http://www.cisco.com/en/US/products/ps7273/tsd_products_support_series_home.html

Related Topic

Telephony Features Available for the Cisco Unified IP Phone, page 5-1

Configuring Network Parameters Using the Cisco Unified IP Phone

You can configure parameters such as DHCP, TFTP, and IP settings on the phone itself. You can also obtain statistics about a call or firmware versions on the phone.

For more information about configuring features and viewing statistics from the phone, see Configuring Settings on the Cisco Unified IP Phone and see Viewing Model Information, Status, and Statistics on the Cisco Unified IP Phone

Providing Users with Feature Information

If you are a system administrator, you are likely the primary source of information for Cisco Unified IP Phone users in your network or company. To ensure that you distribute the most current feature and procedural information, familiarize yourself with Cisco Unified IP Phone documentation. Make sure to visit the Cisco Unified IP Phone web site:

http://www.cisco.com/en/US/products/hw/phones/ps379/tsd_products_support_series_home.html

From this site, you can view and order various user guides, including wallet cards. For complete ordering information, see Document Conventions, page -3.

In addition to providing users with documentation, it is important to inform them of available Cisco Unified IP Phone features—including features specific to your company or network—and of how to access and customize those features, if appropriate.

For a summary of some of the key information that phone users need their system administrators to provide, see Appendix A, Providing Information to Users.

Understanding Security Features for Cisco Unified IP Phones

Implementing security in the Cisco Unified Communications Manager system prevents identity theft of the phone and Cisco Unified Communications Manager server, prevents data tampering, and prevents call signaling and media stream tampering.

To alleviate these threats, the Cisco Unified Communications network establishes and maintains authenticated and encrypted communication streams between a phone and the server, digitally signs files before they are transferred to a phone and encrypts media streams between Cisco Unified IP phones.

If you configure security-related settings in Cisco Unified Communications Manager Administration, the phone configuration file will contain sensitive information. To ensure the privacy of a configuration file, you must configure it for encryption. For detailed information, refer to Configuring Encrypted Phone Configuration Files in Cisco Unified Communications Manager Security Guide.

Table 1-3 shows where you can find additional information about security in this and other documents.

Table 1-3 Cisco Unified IP Phone Security Topics

Topic	Reference
Detailed explanation of security, including set up, configuration, and troubleshooting information for Cisco Unified Communications Manager and Cisco Unified IP Phones	Refer to the *Troubleshooting Guide for Cisco Unified Communications Manager.*
Security features supported on the Cisco Unified IP Phone	See Overview of Supported Security Features, page 1-11.
Restrictions regarding security features	See Security Restrictions, page 1-18.
Identifying phone calls for which security is implemented	See Identifying Authenticated, Encrypted and Protected Phone Calls, page 1-13.
Transport Layer Security (TLS) connection	• See What Networking Protocols are Used?, page 1-3. • See Understanding Phone Configuration Files, page 2-4.
802.1X authentication for Cisco Unified IP Phones	See these sections: • Supporting 802.1X Authentication on Cisco Unified IP Phones, page 1-16. • Security Configuration Menu, page 4-33. • 802.1X Authentication and Status, page 4-37. • Troubleshooting Cisco Unified IP Phone Security, page 9-8.
Security and the phone startup process	See Understanding the Phone Startup Process, page 2-5.
Security and phone configuration files	See Understanding Phone Configuration Files, page 2-4.
Changing the TFTP Server 1 or TFTP Server 2 option on the phone when security is implemented	See Network Configuration Menu, page 4-5.
Understanding security icons in the Communications Manager 1 through Communications Manager 5 options in the Device Configuration Menu on the phone	See Unified CM Configuration Menu, page 4-18.
Items on the Security Configuration menu that you access from the Device Configuration menu on the phone	See Security Configuration Menu, page 4-27.

Table 1-3 Cisco Unified IP Phone Security Topics (continued)

Topic	Reference
Items on the Security Configuration menu that you access from the Settings menu on the phone	See Security Configuration Menu, page 4-33.
Unlocking the Certificate Trust List (CTL) and Identity Trust List (ITL) files	See Unlocking the CTL and ITL files, page 4-35.
Disabling access to a phone's web pages	See Disabling and Enabling Web Page Access, page 8-3.
Deleting the CTL and ITL files from the phone	See Resetting or Restoring the Cisco Unified IP Phone, page 9-12.
Resetting or restoring the phone	See Resetting or Restoring the Cisco Unified IP Phone, page 9-12.
Extension Mobility HTTPS Support	See What Networking Protocols are Used?, page 1-3.
802.1X Authentication for Cisco Unified IP Phones	See these sections: • Supporting 802.1X Authentication on Cisco Unified IP Phones, page 1-16. • 802.1X Authentication and Status, page 4-37. • Troubleshooting Cisco Unified IP Phone Security, page 9-8. • Trust List Menu, page 4-36.

Overview of Supported Security Features

This section provides an overview of the security features that the phone supports. For more information about these features and about Cisco Unified Communications Manager and Cisco Unified IP Phone security, refer to *Cisco Unified Communications Manager Security Guide*.

For information about current security settings on a phone, look at the Security Configuration menus (press the **Applications Menu button** and choose **Settings > Security Configuration** or **Settings > Device Configuration > Security Configuration**). For more information, see Configuring Settings on the Cisco Unified IP Phone.

Note Most security features are available only if a CTL or ITL file or both are installed on the phone. For more information about the CTL and ITL files, refer to *Cisco Unified Communications Manager Security Guide*.

Table 1-4 Overview of Security Features

Feature	Description
Image authentication	Signed binary files (with the extension .sbn) prevent tampering with the firmware image before it is loaded on a phone. Tampering with the image causes a phone to fail the authentication process and reject the new image.
802.1X Authentication	The Cisco Unified IP Phone can use 802.1X authentication to request and gain access to the network. See Supporting 802.1X Authentication on Cisco Unified IP Phones, page 1-16 for more information.

Table 1-4 Overview of Security Features (continued)

Feature	Description
Customer-site certificate installation	Each Cisco Unified IP Phone requires a unique certificate for device authentication. Phones include a manufacturing installed certificate (MIC), but for additional security, you can specify in Cisco Unified Communications Manager Administration that a certificate be installed by using the Certificate Authority Proxy Function (CAPF). Alternatively, you can install an Locally Significant Certificate (LSC) from the Security Configuration menu on the phone. See Configuring Security on the Cisco Unified IP Phone, page 3-12 for more information.
Device authentication	Occurs between the Cisco Unified Communications Manager server and the phone when each entity accepts the certificate of the other entity. Determines whether a secure connection between the phone and a Cisco Unified Communications Manager should occur, and, if necessary, creates a secure signaling path between the entities by using transport layer security (TLS) protocol. Cisco Unified Communications Manager does not register phones configured in authenticated or encrypted mode unless they can be authenticated by the Cisco Unified Communications Manager.
File authentication	Validates digitally signed files that the phone downloads. The phone validates the signature to make sure that file tampering did not occur after the file creation. Files that fail authentication are not written to Flash memory on the phone. The phone rejects such files without further processing.
Signaling Authentication	Uses the TLS protocol to validate that no tampering has occurred to signaling packets during transmission.
Manufacturing installed certificate	Each Cisco Unified IP Phones 7906G and 7911G contains a unique MIC, which is used for device authentication. The MIC is a permanent unique proof of identity for the phone, and allows Cisco Unified Communications Manager to authenticate the phone.
Secure SRST reference	After you configure a SRST reference for security and then reset the dependent devices in Cisco Unified Communications Manager Administration, the TFTP server adds the SRST certificate to the phone cnf.xml file and sends the file to the phone. A secure phone then uses a TLS connection to interact with the SRST-enabled router.
Media encryption	Uses SRTP to ensure that the media streams between supported devices proves secure and that only the intended device receives and reads the data. Includes creating a media master key pair for the devices, delivering the keys to the devices, and securing the delivery of the keys while the keys are in transport.
Signaling Encryption	Ensures that all SCCP and SIP signaling messages that are sent between the device and the Cisco Unified Communications Manager server are encrypted.
CAPF (Certificate Authority Proxy Function)	Implements parts of the certificate generation procedure that are too processing-intensive for the phone, and interacts with the phone for key generation and certificate installation. The CAPF can be configured to request certificates from customer-specified certificate authorities on behalf of the phone, or it can be configured to generate certificates locally.

Table 1-4 Overview of Security Features (continued)

Feature	Description
Optional disabling of the web server functionality for a phone	You can prevent access to a phone's web page, which displays a variety of operational statistics for the phone.
Phone hardening	Additional security options, which you control from Cisco Unified Communications Manager Administration: • Disabling PC port (applies to 7911G only) • Disabling Gratuitous Address Resolution Protocol (GARP) • Disabling PC Voice VLAN access (applies to 7911G only) • Disabling access to the Setting menus, or providing restricted access that allows access to the User Preferences menu and saving volume changes only • Disabling access to web pages for a phone **Note** You can view current settings for the PC Port Disabled, GARP Enabled, and Voice VLAN enabled options by looking at the phone's Security Configuration menu. For more information, see Device Configuration Menu, page 4-17.

Related Topics

- Identifying Authenticated, Encrypted and Protected Phone Calls, page 1-13
- Supporting 802.1X Authentication on Cisco Unified IP Phones, page 1-16
- Security Restrictions, page 1-18
- Device Configuration Menu, page 4-17

Understanding Security Profiles

All Cisco Unified IP Phones that support Cisco Unified Communications Manager 5.0 and later use a security profile, which defines whether the phone is nonsecure, authenticated, or encrypted. For information about configuring the security profile and applying the profile to the phone, refer to the *Cisco Unified Communications Manager Security Guide*.

To view the security mode that is set for the phone, look at the Security Mode setting in the Security Configuration menu. For more information, see Security Configuration Menu.

Related Topics

- Identifying Authenticated, Encrypted and Protected Phone Calls, page 1-13
- Security Restrictions, page 1-18

Identifying Authenticated, Encrypted and Protected Phone Calls

When security is implemented for a phone, you can identify authenticated or encrypted phone calls by icons on the LCD screen on the phone. You can also determine if the connected phone is secure and protected if a security tone plays at the beginning of the call.

In an authenticated call, all devices participating in the establishment of the call are trusted devices, and authenticated by Cisco Unified Communications Manager. When a call in progress is authenticated end-to-end, the call progress icon to the right of the call duration timer in the phone LCD screen changes to the following icon:

In an encrypted call, all devices participating in the establishment of the call are trusted devices, and authenticated by the Cisco Unified Communications Manager. In addition, call signaling and media streams are encrypted. An encrypted call offers a high level of security, providing integrity and privacy to the call. When a call in progress is being encrypted, the call progress icon to the right of the call duration timer in the phone LCD screen changes to the following icon:

Note If the call is routed through a non-IP call leg, for example, PSTN, the call will be nonsecure even though it is encrypted within the IP network and has a lock icon associated with it.

In a protected call, a security tone plays at the beginning of a call to indicate that the other connected phone is also receiving and transmitting encrypted audio and video (if video is involved). If your call is connected to a non-protected phone, the security tone does not play.

Note Protected calling is supported for connections between two phones only. Some features, such as conference calling, shared lines, Extension Mobility, and Join Across Lines are not available when protected calling is configured. Protected calls are not authenticated.

Related Topic

- Understanding Security Features for Cisco Unified IP Phones, page 1-10
- Supporting 802.1X Authentication on Cisco Unified IP Phones, page 1-16
- Security Restrictions, page 1-18

Establishing and Identifying Secure Conference Calls

You can initiate a secure conference call and monitor the security level of participants. A secure conference call is established using this process:

1. A user initiates the conference from a secure phone (encrypted or authenticated security mode).
2. Cisco Unified Communications Manager assigns a secure conference bridge to the call.
3. As participants are added, Cisco Unified Communications Manager verifies the security mode of each phone (encrypted or authenticated) and maintains the secure level for the conference.
4. The phone displays the security level of the conference call. A secure conference displays (encrypted) or (authenticated) icon to the right of "Conference" on the phone screen. If icon displays, the conference is not secure.

Note There are interactions, restrictions, and limitations that affect the security level of the conference call depending on the security mode of the participant's phones and the availability of secure conference bridges. See Table 1-5 and Table 1-6 for information about these interactions.

Establishing and Identifying Protected Calls

A protected call is established when your phone, and the phone on the other end, is configured for protected calling. The other phone can be in the same Cisco IP network, or on a network outside the IP network. Protected calls can only be made between two phones. Conference calls and other multiple-line calls are not supported.

A protected call is established using this process:

1. A user initiates the call from a protected phone (protected security mode).
2. The phone displays the icon (encrypted) on the phone screen. This icon indicates that the phone is configured for secure (encrypted) calls, but this does not mean that the other connected phone is also protected.
3. A security tone plays if the call is connected to another protected phone, indicating that both ends of the conversation are encrypted and protected. If the call is connected to a non-protected phone, then the secure tone is not played.

Note Protected calling is supported for conversations between two phones. Some features, such as conference calling, shared lines, Cisco Extension Mobility, and Join Across Lines are not available when protected calling is configured.

Call Security Interactions and Restrictions

Cisco Unified Communications Manager checks the phone security status when conferences are established and changes the security indication for the conference or blocks the completion of the call to maintain integrity and also security in the system. Table 1-5 provides information about changes to call security levels when using Barge.

Table 1-5 Call Security Interactions When Using Barge

Initiator's Phone Security Level	Feature Used	Call Security Level	Results of Action
Non-secure	Barge	Encrypted call	Call barged and identified as non-secure call
Secure (encrypted)	Barge	Authenticated call	Call barged and identified as authenticated call
Secure (authenticated)	Barge	Encrypted call	Call barged and identified as authenticated call
Non-secure	Barge	Authenticated call	Call barged and identified as non-secure call

Table 1-6 provides information about changes to conference security levels depending on the initiator's phone security level, the security levels of participants, and the availability of secure conference bridges.

Table 1-6 Security Restrictions with Conference Calls

Initiator's Phone Security Level	Feature Used	Security Level of Participants	Results of Action
Non-secure	Conference	Encrypted or authenticated	Non-secure conference bridge Non-secure conference
Secure (encrypted or authenticated)	Conference	At least one member is non-secure	Secure conference bridge Non-secure conference
Secure (encrypted)	Conference	All participants are encrypted	Secure conference bridge Secure encrypted level conference
Secure (authenticated)	Conference	All participants are encrypted or authenticated	Secure conference bridge Secure authenticated level conference
Non-secure	Conference	Encrypted or authenticated	Only secure conference bridge is available and used Non-secure conference
Secure (encrypted or authenticated)	Conference	Encrypted or authenticated	Only non-secure conference bridge is available and used Non-secure conference
Secure (encrypted or authenticated)	Conference	Secure or encrypted	Conference remains secure When one participant tries to Hold the call with MOH, the MOH does not play.
Secure (encrypted)	Join	Encrypted or authenticated	Secure conference bridge Conference remains secure (encrypted or authenticated)
Non-secure	cBarge	All participants are encrypted	Secure conference bridge Conference changes to non-secure
Non-secure	MeetMe	Minimum security level is encrypted	Initiator receives message "Does not meet Security Level", call rejected.
Secure (encrypted)	MeetMe	Minimum security level is authenticated	Secure conference bridge Conference accepts encrypted and authenticated calls
Secure (encrypted)	MeetMe	Minimum security level is non-secure	Only secure conference bridge available and used Conference accepts all calls

Supporting 802.1X Authentication on Cisco Unified IP Phones

These sections provide information about 802.1X support on the Cisco Unified IP Phones:

- Overview, page 1-17
- Required Network Components, page 1-17
- Best Practices—Requirements and Recommendations, page 1-17

Overview

Cisco Unified IP phones and Cisco Catalyst switches have traditionally used Cisco Discovery Protocol (CDP) to identify each other and to determine parameters such as VLAN allocation and inline power requirements. However, CDP is not used to identify any locally attached PCs; therefore, Cisco Unified IP Phones provide an EAPOL pass-through mechanism, whereby a PC locally attached to the IP phone may pass through EAPOL messages to the 802.1X authenticator in the LAN switch. This capability prevents the IP phone from having to act as the authenticator, yet allows the LAN switch to authenticate a data end point prior to accessing the network.

In conjunction with the EAPOL pass-through mechanism, Cisco Unified IP Phones provide a proxy EAPOL-Logoff mechanism. If the locally attached PC is disconnected from the IP phone, the LAN switch would not see the physical link fail, because the link between the LAN switch and the IP phone is maintained. To avoid compromising network integrity, the IP phone sends an EAPOL-Logoff message to the switch on behalf of the downstream PC, which triggers the LAN switch to clear the authentication entry for the downstream PC.

The Cisco Unified IP phones contain an 802.1X supplicant in addition to the EAPOL pass-through mechanism. This supplicant allows network administrators to control the connectivity of IP phones to the LAN switch ports. The current release of the phone 802.1X supplicant uses the EAP-FAST, EAP-TLS, and EAP-MD5 options for network authentication.

Required Network Components

Support for 802.1X authentication on Cisco Unified IP Phones requires several components, including:

- Cisco Unified IP Phone—The phone acts as the 802.1X supplicant, which initiates the request to access the network.
- Cisco Secure Access Control Server (ACS) (or other third-party authentication server)—The authentication server and the phone must both be configured with a shared secret that is used to authenticate the phone.
- Cisco Catalyst Switch (or other third-party switch)—The switch must support 802.1X so it can act as the *authenticator* and pass the messages between the phone and the authentication server. When the exchange is completed, the switch grants or denies the phone access to the network.

Best Practices—Requirements and Recommendations

- Enable 802.1X Authentication—If you want to use the 802.1X standard to authenticate Cisco Unified IP Phones, make sure that you have properly configured the other components before enabling it on the phone. See 802.1X Authentication and Status, page 4-37for more information.
- Configure PC Port—The 802.1X standard does not take into account the use of VLANs and thus recommends that only a single device be authenticated to a specific switch port. However, some switches (including Cisco Catalyst switches) support multi-domain authentication. The switch configuration determines whether you can connect a PC to the phone PC port.
  - Enabled—If you are using a switch that supports multi-domain authentication, you can enable the PC port and connect a PC to it. In this case, Cisco Unified IP Phones support proxy EAPOL-Logoff to monitor the authentication exchanges between the switch and the attached PC. For more information about IEEE 802.1X support on the Cisco Catalyst switches, refer to the Cisco Catalyst switch configuration guides at:

    http://www.cisco.com/en/US/products/hw/switches/ps708/tsd_products_support_series_home.html

  - Disabled—If the switch does not support multiple 802.1X-compliant devices on the same port, you should disable the PC Port when 802.1X authentication is enabled. See Security Configuration Menu, page 4-27 for more information. If you do not disable this port and subsequently attempt to attach a PC to it, the switch will deny network access to both the phone and the PC.

- Configure Voice VLAN—Because the 802.1X standard does not account for VLANs, you should configure this setting based on the switch support.
  - Enabled—If you are using a switch that supports multi-domain authentication, you can continue to use the voice VLAN.
  - Disabled—If the switch does not support multi-domain authentication, disable the Voice VLAN and consider assigning the port to the native VLAN. See Security Configuration Menu, page 4-27 for more information.
- Enter MD5 Shared Secret—If you disable 802.1X authentication or perform a factory reset on the phone, the previously configured MD5 shared secret is deleted. See 802.1X Authentication and Status, page 4-37 for more information.

Security Restrictions

A user cannot barge into an encrypted call if the phone that is used to barge is not configured for encryption. When barge fails in this case, a reorder tone (fast busy tone) plays on the barge initiator's phone.

If the initiator phone is configured for encryption, the barge initiator can barge into an authenticated or nonsecure call from the encrypted phone. After the barge occurs, Cisco Unified Communications Manager classifies the call as nonsecure.

If the initiator phone is configured for encryption, the barge initiator can barge into an encrypted call, and the phone indicates that the call is encrypted.

A user can barge into an authenticated call, even if the phone that is used to barge is nonsecure. The authentication icon continues to appear on the authenticated devices in the call, even if the initiator phone does not support security.

Overview of Configuring and Installing Cisco Unified IP Phones

When deploying a new Unified Communications system, system administrators and network administrators must complete several initial configuration tasks to prepare the network for Unified Communications service. For information and a checklist for setting up and configuring a complete Cisco Unified Communications network, refer to System Configuration Overview in the *Cisco Unified Communications Manager System Guide*.

After you have set up the Unified Communications system and configured system-wide features in Cisco Unified Communications Manager, you can add IP phones to the system.

The following topics provide an overview of procedures for adding Cisco Unified IP Phones to your network:

- Configuring Cisco Unified IP Phones in Cisco Unified Communications Manager, page 1-19
- Installing Cisco Unified IP Phones, page 1-22

Configuring Cisco Unified IP Phones in Cisco Unified Communications Manager

To add phones to the Cisco Unified Communications Manager database, you can use:

- Auto-registration
- Cisco Unified Communications Manager Administration
- Bulk Administration Tool (BAT)
- BAT and the Tool for Auto-Registered Phones Support (TAPS)

For more information about these choices, see Adding Phones to the Cisco Unified Communications Manager Database, page 2-7.

For general information about configuring phones in Cisco Unified Communications Manager, refer to the following documentation:

- Cisco Unified IP Phone, *Cisco Unified Communications Manager System Guide*
- Configuring Cisco Unified IP Phone Configuration, *Cisco Unified Communications Manager Administration Guide*
- Autoregistration, *Cisco Unified Communications Manager Administration Guide*
- *Cisco Unified Communications Manager Bulk Administration Guide*

Checklist for Configuring the Cisco Unified IP Phones 7906G and 7911G in Cisco Unified Communications Manager

Table 1-7 provides an overview and checklist of configuration tasks for the Cisco Unified IP Phones 7906G and 7911G in Cisco Unified Communications Manager. The list presents tasks in a suggested order to guide you through the phone configuration process. Some tasks are optional, depending on your system and user needs. For detailed procedures and information, refer to the sources in the list.

Table 1-7 Checklist for Configuring the Cisco Unified IP Phones 7906G and 7911G in Cisco Unified Communications Manager

Task	Purpose	For More Information
1.	Gather the following information about the phone: • Phone Model • MAC address • Physical location of the phone • Name or user ID of phone user • Device pool • Calling search space and location information (if used) • Number of lines, associated directory numbers (DNs), and partitions to assign to the phone • Cisco Unified Communications Manager user to associate with the phone • Phone usage information that affects phone button template, softkey template, phone features, IP Phone services, or phone applications Provides list of configuration requirements for setting up phones. Identifies preliminary configuration that you need to perform before configuring individual phones, such as phone button templates or softkey templates.	Refer to the *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phone. See Telephony Features Available for the Cisco Unified IP Phone, page 5-1

Table 1-7 Checklist for Configuring the Cisco Unified IP Phones 7906G and 7911G in Cisco Unified Communications Manager (continued)

Task	Purpose	For More Information
2.	Customize phone button templates (if required). Adds Privacy feature to meet user needs. You must specify a service URL with an IPv4 address.	Refer to the *Cisco Unified Communications Manager Administration Guide*, Phone Button Template Configuration. See Modifying Phone Button Templates, page 5-18.
3.	Add and configure the phone by completing these required fields in the Phone Configuration window: • Phone type • MAC address • Device pool • Button template • Product Specific Configuration • Softkey template (if customized) Adds the device with its default settings to the Cisco Unified Communications Manager database.	Refer to the *Cisco Unified Communications Manager Administration Guide*, Cisco Unified IP Phone Configuration chapter. For information about Product Specific Configuration fields, refer to ? Button Help in the Phone Configuration window. **Note** If you want to add both the phone and user to the Cisco Unified Communications Manager database at the same time, refer to *Cisco Unified Communications Manager Administration Guide*, User/Phone Configurations.
4.	Add and configure the directory number on the phone by completing these required fields in the Directory Number Configuration window. • Directory number • Multiple Calls and Call Waiting • Call Forwarding and Pickup (if used) • Voice Messaging (if used) Adds primary and secondary directory numbers and features associated with directory numbers to the phone.	Refer to the *Cisco Unified Communications Manager Administration Guide*: • Directory Number Configuration • See Telephony Features Available for the Cisco Unified IP Phone, page 5-1.
5.	Customize softkey templates (optional). Adds, deletes, or changes order of softkey features that display on the user's phone to meet feature usage needs.	Refer to the *Cisco Unified Communications Manager Administration Guide*, Softkey Template Configuration. See Configuring Softkey Templates, page 5-19.
6.	Configure speed-dial buttons and assign speed-dial numbers (optional). Adds speed-dial numbers. **Note** Users can change speed-dial settings on their phones with Cisco Unified CM User Options.	Refer to the *Cisco Unified Communications Manager Administration Guide*, Cisco Unified IP Phone Configuration chapter, Configuring Speed-Dial Buttons section.
7.	Configure Cisco Unified IP Phone services and assign services (optional). Provides IP Phone services. **Note** Users can add or change services on their phones by using the Cisco Unified CM User Options. **Note** You must specify a service URL with an IPv4 address.	Refer to the *Cisco Unified Communications Manager Administration Guide*, Cisco Unified IP Phone Services Configuration chapter. See Setting Up Services, page 5-20.

Table 1-7 Checklist for Configuring the Cisco Unified IP Phones 7906G and 7911G in Cisco Unified Communications Manager (continued)

Task	Purpose	For More Information
8.	Assign services to phone buttons (optional). Provides single button access to an IP phone service or URL.	Refer to *Cisco Unified Communications Manager Administration Guide*, Cisco Unified IP Phone Configuration, Adding a Cisco Unified IP Phone Service to a Phone Button.
9.	Add user information by configuring required fields (optional). • Name (last) • User ID • Password (for User Options web pages) • PIN (for use with Extension Mobility Adds user information to the global directory for Cisco Unified Communications Manager. **Note** To search for a user in the Corporate Directory, you must add users to Cisco Unified Communications Manager.	Refer to the *Cisco Unified Communications Manager Administration Guide*, End User Configuration. See Adding Users to Cisco Unified Communications Manager, page 5-21. **Note** If your company uses a a Lightweight Directory Access Protocol (LDAP) directory to store information on users, you install and configure Cisco Unified Communications to use your existing LDAP directory, refer to Configuring Corporate and Personal Directories, page 5-16. **Note** If you want to add both the phone and user to the Cisco Unified Communications Manager database at the same time, refer to *Cisco Unified Communications Manager Administration Guide*, User/Phone Configurations.
10.	Add a user to a user group. Assigns users a common list of roles and permissions that apply to all users in a user group. Administrators can manage user groups, roles, and permissions to control the level of access (and, therefore, the level of security) for system users. For example, you must add users to the standard Cisco CCM End Users group so users can access Cisco Unified CM User Options.	Refer to the *Cisco Unified Communications Manager Administration Guide*, User Group Configuration, Adding Users to a User Group.
11.	Associate a user with a phone (optional). Provides users with control over their phone such as forwarding calls or adding speed-dial numbers or services. **Note** Some phones, such as those in conference rooms, do not have an associated user.	Refer to the *Cisco Unified Communications Manager Administration Guide*, End User Configuration, Associating Devices to a User.

Installing Cisco Unified IP Phones

After you have added the phones to the Cisco Unified Communications Manager database, you can complete the phone installation. You (or the phone users) can install the phone at the users's location. The Cisco Unified IP Phone Installation Guide, which is available on Cisco.com, provides directions for connecting the phone footstand, handset, cables, and other accessories.

Note Before you install a phone, even if it is new, upgrade the phone to the current firmware image. For information about upgrading your phone, see the Readme file for your phone model located at:

http://www.cisco.com/cgi-bin/tablebuild.pl/ip-7900ser

After the phone is connected to the network, the phone startup process begins, and the phone registers with Cisco Unified Communications Manager. To finish installing the phone, configure the network settings on the phone depending on whether you enable or disable DHCP service.

If you used auto-registration, you need to update the specific configuration information for the phone such as associating the phone with a user, changing the button table, or directory number.

Checklist for Installing the Cisco Unified IP Phones 7906G and 7911G

Table 1-8 provides an overview and checklist of installation tasks for the Cisco Unified IP Phone 7906G and 7911G. The list presents tasks in a suggested order to guide you through the phone installation process. Some tasks are optional, depending on your system and user needs. For detailed procedures and information, refer to the sources in the list.

Table 1-8 Checklist for Installing the Cisco Unified IP Phones 7906G and 7911G

Task	Purpose	For More Information
1.	Choose the power source for the phone: – Power over Ethernet (PoE) – External power supply Determines how the phone receives power.	See Providing Power to the Cisco Unified IP Phone 7906G and 7911G, page 2-3.
2.	Assemble the phone, adjust phone placement, and connect the network cable. Locates and installs the phone in the network.	See Installing the Cisco Unified IP Phone, page 3-5.
3.	Monitor the phone startup process. Verifies that phone is configured properly.	See Verifying the Phone Startup Process, page 3-11.

Table 1-8 Checklist for Installing the Cisco Unified IP Phones 7906G and 7911G (continued)

Task	Purpose	For More Information
4.	If you are configuring the network settings on the phone for an IPv4 network, you can set up an IP address for the phone by using of the following methods: • Using DHCP—To enable DHCP and allow the DHCP server to automatically assign an IP address to the Cisco Unified IP Phone and direct the phone to a TFTP server, choose **Settings > Network Configuration> IPv4 Configuration** and: – To enable DHCP, set DHCP Enabled to **Yes.** DHCP is enabled by default. – To use an alternate TFTP server, set Alternate TFTP Server to **Yes,** and enter the IP address for the TFTP Server. **Note** Consult with the network administrator if you need to assign an alternative TFTP server instead of using the TFTP server assigned by DHCP. • Without DHCP—You must configure the IP address, subnet mask, TFTP server, and default router locally on the phone, choose **Settings > Network Configuration> IPv4 Configuration**: To disable DHCP and manually set an IP address: **a.** To disable DHCP, set DHCP Enabled to **No**. **b.** Enter the static IP address for phone. **c.** Enter the subnet mask. **d.** Enter the default router IP addresses. **e.** Set Alternate TFTP Server to **Yes, and** enter the IP address for TFTP Server 1. **f.** Choose **Settings > Network Configuration** and enter the domain name where the phone resides. **Note** The Cisco Unified IP Phone supports having both IPv4 and an IPv6 address concurrently. You can configure Cisco Unified Communications Manager to support IPv4 addresses only, IPv6 addresses only, or support both IPv4/IPv6 addresses.	See Configuring Startup Network Settings, page 3-11. See Network Configuration Menu, page 4-5.

Table 1-8 Checklist for Installing the Cisco Unified IP Phones 7906G and 7911G (continued)

Task	Purpose	For More Information
5.	If you are configuring the network settings on the phone for an IPv6 network, you can set up an IP address for the phone by using of the following methods: • Using DHCPv6—To enable DHCPv6 and allow the DHCPv6 server to automatically assign an IP address to the Cisco Unified IP Phone and direct the phone to a TFTP server, choose **Settings > Network Configuration> IPv6 Configuration** and configure the following: – To enable DHCPv6, set DHCPv6 to **Yes.** DHCPv6 is enabled by default. – To use an alternate TFTP server, set IPv6 Alternate TFTP Server to **Yes**, and enter the IP address for IPv6 TFTP Server 1. **Note** Consult with the network administrator if you need to assign an alternate TFTP server instead of using the TFTP server assigned by DHCP. • Without DHCP—You must configure the IP address, subnet mask and TFTP server locally on the phone, choose **Settings > Network Configuration> IPv6 Configuration**: To disable DHCP and manually set an IP address: **a.** To disable DHCPv6, set DHCPv6 to **No**. **b.** Enter the static IP address for phone. **c.** Enter the IPv6 prefix length. **d.** Set IPv6 Alternate TFTP Server to **Yes**, and enter IP address for IPv6 TFTP Server 1. **e.** Choose **Settings > Network Configuration** and enter the domain name where the phone resides. **Note** The Cisco Unified IP Phone supports having both IPv4 and an IPv6 address concurrently. You can configure Cisco Unified Communications Manager to support IPv4 devices only, IPv6 devices only, or to support both IPv4 and IPv6 devices concurrently.	See Configuring Startup Network Settings, page 3-11. See Network Configuration Menu, page 4-5.
6.	Set up security on the phone. Provides protection against data tampering threats and identity theft of phones.	See Configuring Security on the Cisco Unified IP Phone, page 3-12.
7.	Make calls with the Cisco Unified IP Phone. Verifies that the phone and features work correctly.	Refer to the *Cisco Unified IP Phones 7906G and 7911G Guide*.
8.	Provide information to end users about how to use their phones and how to configure their phone options. Ensures that users have adequate information to successfully use their Cisco Unified IP Phones.	See Appendix A, Providing Information to Users.

CHAPTER 2

Preparing to Install the Cisco Unified IP Phone on Your Network

Cisco Unified IP Phones enable you to communicate by using voice over a data network. To provide this capability, the IP Phones depend upon and interact with several other key Cisco Unified Communications and network components, including Cisco Unified Communications Manager, DNS and DHCP servers, TFTP servers, and media resources.

This chapter provides an overview of the interaction between the Cisco Unified IP Phone 7906G and 7911G and other key components of the Voice-over-IP (VoIP) network, and focuses on the interactions between the Cisco Unified IP Phone 7906G and 7911G and Cisco Unified Communications Manager, TFTP server, and switches. It includes these topics:

- Understanding Interactions with Other Cisco Unified Communications Products, page 2-1
- Understanding the Phone Startup Process, page 2-5
- Providing Power to the Cisco Unified IP Phone 7906G and 7911G, page 2-3
- Understanding Phone Configuration Files, page 2-4
- Adding Phones to the Cisco Unified Communications Manager Database, page 2-7
- Using Cisco Unified IP Phones with Different Protocols, page 2-10
- Determining the MAC Address of a Cisco Unified IP Phone, page 2-11

Understanding Interactions with Other Cisco Unified Communications Products

To function in the Unified Communications network, the Cisco Unified IP Phone must be connected to a networking device, such as a Cisco Catalyst switch. You must also register the Cisco Unified IP Phone with a Cisco Unified Communications Manager system before sending and receiving calls.

This section includes these topics:

- Understanding How the Cisco Unified IP Phone Interacts with Cisco Unified Communications Manager, page 2-2
- Understanding How the Cisco Unified IP Phone Interacts with the VLAN, page 2-2

Understanding How the Cisco Unified IP Phone Interacts with Cisco Unified Communications Manager

Cisco Unified Communications Manager is an open and industry-standard call processing system. Cisco Unified Communications Manager software sets up and tears down calls between phones, integrating traditional PBX functionality with the corporate IP network. Cisco Unified Communications Manager manages the components of the Cisco Unified Communications system—the phones, the access gateways, and the resources necessary for features such as call conferencing and route planning. Cisco Unified Communications Manager also provides authentication and encryption if configured for the communications system.

For information about configuring Cisco Unified Communications Manager to work with the IP devices described in this chapter, refer to Cisco Unified Communications Manager Administration Guide, *Cisco Unified Communications Manager System Guide*, and *Cisco Unified Communications Manager Security Guide*.

For an overview of security functionality for the Cisco Unified IP Phone, see Overview of Supported Security Features, page 1-11.

Note If the Cisco Unified IP Phone model that you want to configure does not appear in the Phone Type drop-down list in Cisco Unified Communications Manager Administration, go to the following URL and install the latest support patch for your version of Cisco Unified Communications Manager: http://www.cisco.com/kobayashi/sw-center/sw-voice.shtml

Related Topic

- Telephony Features Available for the Cisco Unified IP Phone, page 5-1

Understanding How the Cisco Unified IP Phone Interacts with the VLAN

The Cisco Unified IP Phone 7911G has an internal Ethernet switch, which enables forwarding of packets to the phone and to the network port and access port on the back of the phone. The Cisco Unified IP Phone 7906G has an Ethernet port, which enables forwarding of packets to the phone and to the network port.

If a computer is connected to the access port (Cisco Unified IP Phone 7911G), the computer and the phone share the same physical link to the switch and the same port on the switch. This shared physical link affects the VLAN configuration on the network in the following ways:

- Although current VLANs might be configured on an IP subnet basis, additional IP addresses may not be available to assign the phone to the same subnet as other devices that connect to the same port.
- Data traffic present on the data/native VLAN may reduce the quality of Voice-over-IP traffic.
- Network security may necessitate the isolation of the VLAN voice traffic from the VLAN data traffic.

You can resolve these issues by isolating the voice traffic onto a separate VLAN, so that the switch port to which the phone is connected uses separate VLANs for the following types of traffic:

- Voice traffic to and from the IP phone (auxiliary VLAN, on the Cisco Catalyst 6000 series, for example)
- Data traffic to and from the PC connected to the switch through the access port of the IP phone (native VLAN, 7911G only)

Isolating the phones on a separate, auxiliary VLAN improves the quality of the voice traffic and allows a large number of phones to be added to an existing network in which there are not enough IP addresses for each phone.

For more information, refer to the documentation included with a Cisco switch. You can also access related documentation at this URL:

http://www.cisco.com/en/US/products/hw/switches/tsd_products_support_category_home.html

Related Topics

- Understanding the Phone Startup Process, page 2-5
- Network Configuration Menu, page 4-5

Providing Power to the Cisco Unified IP Phone 7906G and 7911G

The Cisco Unified IP Phone 7906G and 7911G can be powered with external power or with Power over Ethernet (PoE). External power is provided through a separate power supply. PoE is provided by a switch through the Ethernet cable attached to a phone.

These sections provide more information about powering a phone:

- Power Guidelines, page 2-3
- Power Outage, page 2-4
- Obtaining Additional Information about Power, page 2-4

Power Guidelines

Table 2-1 provides guidelines that apply to external power and to PoE power for the Cisco Unified IP Phone 7906G and 7911G.

Table 2-1 Guidelines for Powering the Cisco Unified IP Phone 7906G and 7911G

Power Type	Guidelines
External power—Provided through a Cisco external power supply.	The Cisco Unified IP Phone Series use the CP-PWR-CUBE-3 power supply.
External power—Provided through the Cisco Unified IP Phone Power Injector.	The Cisco Unified IP Phone Power Injector may be used with any Cisco Unified IP Phone. Functioning as a midspan device, the injector delivers inline power to the attached phone. The Cisco Unified IP Phone Power Injector is connected between a switch port and the IP Phone, and supports a maximum cable length of 100m between the unpowered switch and the phone.
PoE power—Provided by a switch through the Ethernet cable attached to the phone.	• The Cisco Unified IP Phone 7906G and 7911G support both Cisco inline power and IEEE 802.3af Power over Ethernet. • To ensure uninterruptible operation of the phone, make sure that the switch has a backup power supply. • Make sure that the CatOS or IOS version running on your switch supports your intended phone deployment. Refer to the documentation for your switch for operating system version information.

Power Outage

Your accessibility to emergency service through the phone is dependent on the phone being powered. If there is an interruption in the power supply, Service and Emergency Calling Service dialing will not function until power is restored. In the case of a power failure or disruption, you may need to reset or reconfigure equipment before using the Service or Emergency Calling Service dialing.

Obtaining Additional Information about Power

For related information about power, refer to the documents shown in Table 2-2. These documents provide information about these topics:

- Cisco switches that work with the Cisco Unified IP Phones 7906G and 7911G
- The Cisco IOS releases that support bidirectional power negotiation
- Other requirements and restrictions regarding power

Table 2-2 Related Documentation for Power

Document Topics	URL
Cisco Unified IP Phone Power Injector	http://www.cisco.com/en/US/products/hw/phones/ps379/prod_installation_guides_list.html
PoE Solutions	http://www.cisco.com/en/US/netsol/ns340/ns394/ns147/ns412/networking_solutions_package.html
Cisco Catalyst Switches	http://www.cisco.com/en/US/products/hw/switches/ps708/tsd_products_support_series_home.html
Integrated Service Routers	http://www.cisco.com/en/US/products/hw/routers/index.html
Cisco IOS Software	http://www.cisco.com/en/US/products/sw/iosswrel/products_ios_cisco_ios_software_category_home.html

Understanding Phone Configuration Files

Configuration files for a phone are stored on the TFTP server and define parameters for connecting to Cisco Unified Communications Manager. In general, any time you make a change in Cisco Unified Communications Manager that requires the phone to be reset, a change is automatically made to the phone's configuration file.

Configuration files also contain information about which image load the phone should be running. If this image load differs from the one currently loaded on a phone, the phone contacts the TFTP server to request the required load files. These files are digitally signed to ensure the authenticity of the files' source.

In addition, if the device security mode in the configuration file is set to Authenticated and the CTL file on the phone has a valid certificate for Cisco Unified Communications Manager, the phone establishes a TLS connection to Cisco Unified Communications Manager Administration. Otherwise, the phone establishes a TCP connection.

Note Cisco Extension Mobility Cross Cluster is an exception, in that the phone permits a TLS connection to Cisco Unified Communications Manager for secure signaling even without the CTL file.

If you configure security-related settings in Cisco Unified Communications Manager Administration, the phone configuration file will contain sensitive information. To ensure the privacy of a configuration file, you must configure it for encryption. For detailed information, refer to Configuring Encrypted Phone Configuration Files in *Cisco Unified Communications Manager Security Guide*.

A phone requests a configuration file whenever it resets and registers with Cisco Unified Communications Manager.

A phone accesses a default configuration file named XmlDefault.cnf.xml only when the phone has not received a valid Trust List file containing a certificate assigned to the Cisco Unified Communications Manager and TFTP.

If auto registration is not enabled and you did not add the phone to the Cisco Unified Communications Manager database, the phone does not attempt to register with Cisco Unified Communications Manager. The phone continually displays the "Configuring IP" message until you either enable auto-registration or add the phone to the Cisco Unified Communications Manager database.

If the phone has registered before, the phone will access the configuration file named SEP*mac_address*.cnf.xml, where *mac_address* is the MAC address of the phone. For more information about how the phone interacts with the TFTP server, refer to *Cisco Unified Communications Manager System Guide*, Cisco TFTP.

If you configure security-related settings in Cisco Unified Communications Manager Administration, the phone configuration file will contain sensitive information. To ensure the privacy of a configuration file, you must configure it for encryption. For more information, refer to Configuring Encrypted Phone Configuration Files in *Cisco Unified Communications Manager Security Guide*.

SIP Dial Rules

For Cisco Unified IP Phones running under SIP, the administrator uses dial rules to configure SIP phone dial plans. These dial plans must be associated with a SIP phone device to enable dial plans to be sent to the configuration file. If the administrator does not configure a SIP phone dial plan, the phone does not display any indication of a dial plan. In this case, you must press the Dial softkey, unless the phone supports key press markup language (KPML).

For more information on configuring SIP dial rules, refer to the *Cisco Unified Communications Manager Administration Guide*.

Understanding the Phone Startup Process

When connecting to the VoIP network, the Cisco IP Phone goes through a standard startup process, as described in Table 2-3. Depending on your specific network configuration, not all of these steps may occur on your Cisco Unified IP Phone.

Table 2-3 Cisco Unified IP Phone Startup Process

Task	Purpose	Related Topics
1.	Obtaining Power from the Switch. If a phone is not using external power, the switch provides in-line power through the Ethernet cable attached to the phone.	See Providing Power to the Cisco Unified IP Phone 7906G and 7911G, page 2-3. See Resolving Startup Problems, page 9-1.
2.	The Cisco IP Phone has non-volatile Flash memory in which it stores firmware images and user-defined preferences. At startup, the phone runs a bootstrap loader that loads a phone image stored in Flash memory. Using this image, the phone initializes its software and hardware.	See Resolving Startup Problems, page 9-1.
3.	Configuring VLAN. If the Cisco IP Phone is connected to a Cisco switch, the switch next informs the phone of the voice VLAN defined on the switch port. The phone needs to know its VLAN membership before it can proceed with the Dynamic Host Configuration Protocol (DHCP) request for an IP address. If a third-party switch is used and VLANs are configured, the VLAN on the phone must be manually configured.	See Network Configuration Menu, page 4-5. See Resolving Startup Problems, page 9-1.
4.	Obtaining an IP Address. If the Cisco IP Phone is using DHCP to obtain an IP address, the phone queries the DHCP server to obtain one. If you are not using DHCP in your network, you must assign static IP addresses to each phone locally. In addition to assigning an IP address, the DHCP server directs the Cisco Unified IP Phone to a TFTP Server. If the phone has a statically defined IP address, you must configure the TFTP server locally on the phone; the phone then contacts the TFTP server directly. **Note** You can also assign an alternative TFTP server to use instead of the one assigned by DHCP.	See Network Configuration Menu, page 4-5. See Resolving Startup Problems, page 9-1.
5.	Accessing a TFTP Server.	See Network Configuration Menu, page 4-5. See Resolving Startup Problems, page 9-1.
6.	Requesting the CTL file. The TFTP server stores the CTL file. This file contains the certificates necessary for establishing a secure connection between the phone and Cisco Unified Communications Manager.	Refer to the *Cisco Unified Communications Manager Security Guide*, Configuring the Cisco CTL Client.
7.	Requesting the ITL file The phone requests the ITL file after it requests the CTL file. The ITL file contains the certificates of the entities that the phone can trust. The certificates are used for authenticating a secure connection with the servers or to authenticating a digital signature signed by the servers.	Refer to the *Cisco Unified Communications Manager Security Guide*, Security by Default.

Adding Phones to the Cisco Unified Communications Manager Database

Before installing the Cisco Unified IP phone, you must choose a method for adding phones to the Cisco Unified Communications Manager database. These sections describe the methods:

- Adding Phones with Auto-Registration, page 2-7
- Adding Phones with Auto-Registration and TAPS, page 2-8
- Adding Phones with Cisco Unified Communications Manager Administration, page 2-9
- Adding Phones with BAT, page 2-9

Table 2-4 provides an overview of these methods for adding phones to the Cisco Unified Communications Manager database.

Table 2-4 Methods for Adding Phones to the Cisco Unified Communications Manager Database

Method	Requires MAC Address?	Notes
Auto-registration	No	• Results in automatic assignment of directory numbers. • Not available when security or encryption is enabled.
Auto-registration with TAPS	No	Requires auto-registration and the Bulk Administration Tool (BAT); updates the Cisco Unified Communications Manager database with the MAC address and DNs for the device when user calls TAPS from the phone.
Using the Cisco Unified Communications Manager Administration	Yes	Requires phones to be added individually
Using BAT	Yes	• Can add groups of same model of phone. • Can schedule when phones are added to the Cisco Unified Communications Manager database.

Adding Phones with Auto-Registration

By enabling auto-registration before you begin installing phones, you can:

- Add phones without first gathering MAC addresses from the phones.
- Automatically add a Cisco Unified IP Phone to the Cisco Unified Communications Manager database when you physically connect the phone to your IP telephony network. During auto-registration, Cisco Unified Communications Manager assigns the next available sequential directory number to the phone.
- Quickly enter phones into the Cisco Unified Communications Manager database and modify any settings, such as the directory numbers, from Cisco Unified Communications Manager.
- Move auto-registered phones to new locations and assign them to different device pools without affecting their directory numbers.

Note Cisco recommends you use auto-registration to add less than 100 phones to your network. To add more than 100 phones to your network, use the Bulk Administration Tool (BAT). See Adding Phones with BAT, page 2-9.

Auto-registration is disabled by default. In some cases, you may not want to use auto-registration; for example, if you want to assign a specific directory number to the phone, or if you plan to use secure connection with Cisco Unified Communications Manager, as described in *Cisco Unified Communications Manager Security Guide*. For information about enabling auto-registration, refer to Enabling Auto-Registration in the *Cisco Unified Communications Manager Administration Guide.*

Note When you configure the cluster for mixed mode through the Cisco CTL client, auto-registration is automatically disabled. When you configure the cluster for nonsecure mode through the Cisco CTL client, auto-registration is automatically enabled.

Related Topics

- Adding Phones with Auto-Registration and TAPS, page 2-8
- Adding Phones with Cisco Unified Communications Manager Administration, page 2-9
- Adding Phones with BAT, page 2-9

Adding Phones with Auto-Registration and TAPS

You can add phones with auto-registration and TAPS, the Tool for Auto-Registered Phones Support, without first gathering MAC addresses from phones.

TAPS works with the Bulk Administration Tool (BAT) to update a batch of phones that were already added to the Cisco Unified Communications Manager database with dummy MAC addresses. Use TAPS to update MAC addresses and download pre-defined configurations for phones.

Note Cisco recommends you use auto-registration and TAPS to add less than 100 phones to your network. To add more than 100 phones to your network, use the Bulk Administration Tool (BAT). See Adding Phones with BAT, page 2-9.

To implement TAPS, you or the end-user dial a TAPS directory number and follow voice prompts. When the process is complete, the phone will have downloaded its directory number and other settings, and the phone will be updated in Cisco Unified Communications Manager Administration with the correct MAC address.

Auto-registration must be enabled in Cisco Unified Communications Manager Administration (**System > Cisco Unified CM**) for TAPS to function.

Note When you configure the cluster for mixed mode through the Cisco CTL client, auto-registration is automatically disabled. When you configure the cluster for nonsecure mode through the Cisco CTL client, auto-registration is not automatically enabled.

For detailed instructions about using BAT and TAPS, refer to the Bulk Administration Tool (BAT) chapter *Cisco Unified Communications Manager Administration Guide*.

Related Topics

- Adding Phones with Auto-Registration, page 2-7
- Adding Phones with Cisco Unified Communications Manager Administration, page 2-9
- Adding Phones with BAT, page 2-9

Adding Phones with Cisco Unified Communications Manager Administration

You can add phones individually to the Cisco Unified Communications Manager database using Cisco Unified Communications Manager Administration. To do so, you first need to obtain the MAC address for each phone.

For information about determining a MAC address, see Determining the MAC Address of a Cisco Unified IP Phone, page 2-11.

After you have collected MAC addresses, in Cisco Unified Communications Manager Administration, choose **Device > Phone** and click **Add New** to begin.

For complete instructions and conceptual information about Cisco Unified Communications Manager, refer to *Cisco Unified Communications Manager Administration Guide* and to *Cisco Unified Communications Manager System Guide*.

Related Topics

- Adding Phones with Auto-Registration, page 2-7
- Adding Phones with Auto-Registration and TAPS, page 2-8
- Adding Phones with BAT, page 2-9

Adding Phones with BAT

Cisco Unified Communications Manager Bulk Administration Tool (BAT) enables you to perform batch operations, which includes registration, on multiple phones. To access BAT, choose the **Bulk Administration** drop-down menu in Cisco Unified Communications Manager Administration,

To add phones by using BAT only (not in conjunction with TAPS), you first need to obtain the appropriate MAC address for each phone.

For information about determining a MAC address, see Determining the MAC Address of a Cisco Unified IP Phone, page 2-11.

To add a phone to the Cisco Unified Communications Manager, follow these steps:

Procedure

Step 1 From Cisco Unified Communications Manager, choose **Bulk Administration > Phones > Phone Template**.

Step 2 Click **Add New**.

Step 3 Choose a Phone Type and click **Next**.

Step 4 Enter the details of phone specific parameters like Device Pool, Phone Button Template, Device Security Profile and so on.

Step 5 Click **Save**.

Step 6 From Cisco Unified Communications Manager, choose **Device > Phone > Add New** to add a phone using an already created BAT phone template.

For detailed instructions about using BAT, refer to the Cisco Unified Communications Manager "Bulk Administration Guide. For more information on creation of BAT Phone Templates, see the *Cisco Unified Communications Manager Bulk Administration Guide*, Phone Template.

Related Topics

- Adding Phones with Auto-Registration, page 2-7
- Adding Phones with Auto-Registration and TAPS, page 2-8
- Adding Phones with Cisco Unified Communications Manager Administration, page 2-9

Using Cisco Unified IP Phones with Different Protocols

The Cisco Unified IP Phone can operate with SCCP (Skinny Client Control Protocol) or SIP (Session Initiation Protocol). You can convert a phone that is using one protocol for use with the other protocol.

This section includes these topics:

- Converting a New Phone from SCCP to SIP, page 2-10
- Converting an In-Use Phone from One Protocol to the Other, page 2-11
- Deploying a Phone in an SCCP and SIP Environment, page 2-11

Converting a New Phone from SCCP to SIP

A new, unused phone is set for SCCP by default. To convert this phone to SIP, perform these steps:

Procedure

Step 1 Take one of these actions:

- To auto-register the phone, set the Auto Registration Phone Protocol enterprise parameter in Cisco Unified Communications Manager Administration to SIP.
- To provision the phone by using the Bulk Administration Tool (BAT), choose the appropriate phone model and choose SIP from BAT.
- To provision the phone manually, make the appropriate changes for SIP on the Phone Configuration window in Cisco Unified Communications Manager Administration.

Refer to the *Cisco Unified Communications Manager Administration Guide* for more information about Cisco Unified Communications Manager configuration. Refer to *Bulk Administration Tool Administration Guide* for more information about using BAT.

Step 2 If you are not using DHCP in your network, configure the network parameters for the phone.

See Configuring Startup Network Settings, page 3-11.

Step 3 Save the configuration updates, click **Apply Config**, click **OK** in the Apply Configuration Information window, and have the user power cycle the phone.

Converting an In-Use Phone from One Protocol to the Other

For information on how to convert an in-use phone from one protocol to the other, see the *Cisco Unified Communications Manager Administration Guide*, Cisco Unified IP Phone Configuration, Migration Existing Phone Configuration to a Different Phone.

Deploying a Phone in an SCCP and SIP Environment

To deploy Cisco Unified IP Phones in an environment that includes SCCP and SIP and in which the Cisco Unified Communications Manager Auto-Registration parameter is SCCP, perform these general steps:

1. Set the Cisco Unified Communications Manager auto_registration_protocol parameter to SCCP.

   From Cisco Unified Communications Manager Administration, choose **System > Enterprise Parameters**.
2. Install the phones.
3. Change the Auto Registration Protocol enterprise parameter to SIP.
4. Auto-register the SIP phones.

Determining the MAC Address of a Cisco Unified IP Phone

Several of the procedures that are described in this manual require you to determine the MAC address of a Cisco Unified IP Phone. You can determine the MAC address for a phone in any of these ways:

- From the phone, press the **Applications Menu** button, then choose **Settings > Network Configuration**, and look at the MAC Address field.
- Look at the MAC label on the back of the phone.
- Display the web page for the phone and click the **Device Information** hyperlink.

  For information about accessing the web page, see Accessing the Web Page for a Phone, page 8-1.

CHAPTER 3

Setting Up the Cisco Unified IP Phone

This chapter helps you install the Cisco Unified IP Phones 7906G and 7911G on a Cisco Unified Communications network, and includes these topics:

- Before You Begin, page 3-1
- Understanding the Cisco Unified IP Phone 7906G and 7911G Components, page 3-2
- Installing the Cisco Unified IP Phone, page 3-5
- Mounting the Phone to a Wall, page 3-10
- Verifying the Phone Startup Process, page 3-11
- Configuring Startup Network Settings, page 3-11
- Configuring Security on the Cisco Unified IP Phone, page 3-12

Note Before you install a Cisco Unified IP phone, you must decide how to configure the phone in your network. Then you can install the phone and verify its functionality. For more information, see Preparing to Install the Cisco Unified IP Phone on Your Network, page 2-1.

Before You Begin

Before installing the Cisco Unified IP Phone, review the requirements in these sections:

- Network Requirements, page 3-1
- Cisco Unified Communications Manager Configuration, page 3-2
- Network and Access Ports, page 3-3
- Handset, page 3-3
- Speaker, page 3-3
- Installing the Cisco Unified IP Phone, page 3-5

Network Requirements

For the Cisco Unified IP Phone 7906G and 7911G to successfully operate as a Cisco Unified IP Phone endpoint in your network, your network must meet these requirements:

- Working Voice-over-IP (VoIP) Network

  - VoIP configured on your Cisco routers and gateways
  - Cisco Unified Communications Manager Release 3.3(5) or higher installed in your network and configured to handle call processing

Note The minimum firmware release that must be installed on the phone is 7.2(1).

- IP network that supports DHCP or manual assignment of IP address, gateway, and subnet mask

Note The Cisco Unified IP Phone displays the date and time from Cisco Unified Communications Manager. If the Cisco Unified Communications Manager server is located in a different time zone than the phones, the phones will not display the correct local time.

Cisco Unified Communications Manager Configuration

The Cisco Unified IP Phone requires Cisco Unified Communications Manager to handle call processing. Refer to *Cisco Unified Communications Manager Administration Guide* or context-sensitive help in the Cisco Unified Communications Manager application to ensure that Cisco Unified Communications Manager is set up properly to manage the phone and to properly route and process calls.

If you plan to use auto-registration, verify that it is enabled and properly configured in Cisco Unified Communications Manager Administration before connecting any Cisco Unified IP Phone to the network. For information about enabling and configuring auto-registration, refer to *Cisco Unified Communications Manager Administration Guide*. Also, see Adding Phones to the Cisco Unified Communications Manager Database, page 2-7.

You must use Cisco Unified Communications Manager Administration to configure and assign features to the Cisco Unified IP Phones. See Telephony Features Available for the Cisco Unified IP Phone, page 5-1 for details.

In Cisco Unified Communications Manager Administration, you can add users to the database and associate them with specific phones. In this way, users can access their Cisco Unified CM User Option page to configure items such as call forwarding, speed dialing, and voice messaging system options. See Adding Users to Cisco Unified Communications Manager, page 5-21 for details.

Understanding the Cisco Unified IP Phone 7906G and 7911G Components

The Cisco Unified IP Phone 7906G and 7911G include these components on the phone or as accessories for the phone:

- Network and Access Ports, page 3-3
- Handset, page 3-3
- Speaker, page 3-3
- Headset, page 3-4

Network and Access Ports

The following ports are available on the Cisco Unified IP Phone 7906G and 7911G:

- Network port—Labeled 10/100 SW. Use the network port to connect the phone to the network. You must use a straight-through cable on this port. The phone can also obtain inline power from the Cisco Catalyst switch over this connection. See Providing Power to the Cisco Unified IP Phone 7906G and 7911G, page 2-3 for details.
- Access port (Cisco Unified IP Phone 7911G only)—Labeled 10/100 PC. Use the access port to connect a network device, such as a computer, to the phone. You must use a straight-through cable on this port.

Each port supports 10/100 Mbps half- or full-duplex connections to external devices. The speed and connection type are set through auto-negotiation. You can use either Category 3 or 5 cabling for 10-Mbps connections, but you must use Category 5 for 100 Mbps connections.

See Figure 3-3 for the connection ports available on the back of the Cisco Unified IP Phone 7906G and 7911G.

Handset

The handset is designed especially for use with a Cisco Unified IP Phone. It includes a light strip that indicates incoming calls and voice messages waiting.

To connect a handset to the Cisco Unified IP Phone, plug the cable into the handset and the Handset port on the back of the phone.

Speaker

The Cisco Unified IP Phone 7906G and 7911G include a speaker that you can use to monitor calls. You can enable either the Monitor mode or Group Listen mode to allow users to listen on the speaker.

The speaker is enabled by default. You must disable the speaker by using Cisco Unified Communications Manager Administration. To do so, choose **Device > Phone** and locate the phone you want to modify. In the Phone Configuration window for the phone, check the **Disable Speakerphone** check box.

Monitor Mode

In Monitor mode, users can only listen to a call on the speaker. To speak to the other party on the call, users must pick up the handset.

Monitor mode is enabled by default if the speaker is enabled on Cisco Unified Communications Manager Administration.

From the phone, users can turn on the Monitor function with the **Monitor** softkey, and turn off this function with the **MonOff** softkey or by picking up the handset.

Group Listen Mode

In Group Listen mode, both the handset and speaker can be active at the same time. During a call, one user can talk into the handset while other users can listen over the speaker.

Enabling Group Listen Mode on Cisco Unified Communications Manager

Group Listen mode is disabled by default. To enable this mode, you must do so from the Phone Configuration window in Cisco Unified Communications Manager Administration.

From Cisco Unified Communications Manager Administration, choose **Device > Phone** and locate the phone you want to modify. In the Phone Configuration window for the phone (Product Specific Configuration section), check the **Enable Group Listen** check box.

If Group Listen mode is enabled, the Monitor feature softkeys are not available on the phone.

Activating Group Listen on the Phone

Group Listen softkeys are displayed if Group Listen mode is enabled by the administrator on Cisco Unified Communications Manager. However, these softkeys cannot be configured by using the Cisco Unified Communications Manager softkey template.

- **GListen**—Activates Group Listen on the phone. Displayed when Group Listen mode is enabled by the administrator but not activated on the phone. Once Group Listen is activated on the phone (by pressing **GListen**), users can deactivate it by hanging up the handset or by pressing **GLOff**.
- **GLOff**—Deactivates Group Listen on the phone. Displayed when Group Listen mode is enabled by the administrator and activated on the phone.

Note If Group Listen mode is enabled in Cisco Unified Communications Manager, the **GListen** and **GLOff** softkeys replace the **Monitor** and **MonOff** softkeys on the phone.

Headset

Although Cisco Systems performs internal testing of third-party headsets for use with the Cisco Unified IP Phones, Cisco does not certify or support products from headset or handset vendors.

Cisco recommends the use of good quality external devices, for example, headsets that are screened against unwanted radio frequency (RF) and audio frequency (AF) signals. Depending on the quality of headsets and their proximity to other devices such as cell phones and two-way radios, some audio noise or echo may still occur. An audible hum or buzz may be heard by either the remote party or by both the remote party and the Cisco Unified IP Phone user. Humming or buzzing sounds can be caused by a range of outside sources; for example, electric lights, electric motors, or large PC monitors. See Using External Devices with Your Cisco Unified IP Phone, page 3-5, for more information.

Note In some cases, hum may be reduced or eliminated by using a local power cube or power injector.

These environmental and hardware inconsistencies in the locations where Cisco Unified IP Phones are deployed means that there is not a single headset solution that is optimal for all environments.

Cisco recommends that customers test headsets in their intended environment to determine performance before making a purchasing decision and deploying in mass quantities.

Note The Cisco Unified IP Phone 7906G and 7911G supports wideband headsets.

Audio Quality Subjective to User

Beyond the physical, mechanical and technical performance, the audio portion of a headset must sound good to the user and to the party on the far end. Sound quality is subjective and Cisco cannot guarantee the performance of any headsets. However, a variety of headsets from leading headset manufacturers have been reported to perform well with Cisco Unified IP Phones. See manufacturer's sites for details.

Connecting a Headset

To connect a headset to the Cisco Unified IP Phone, plug it into the RJ-9 Handset port on the back of the phone. Depending on headset manufacturer's recommendations, an external amplifier may be required. Refer to headset manufacturer's product documentation for details.

You can use the headset with all of the features on the Cisco Unified IP Phone, including using the Volume button.

Using External Devices with Your Cisco Unified IP Phone

The following information applies when you use external devices with the Cisco Unified IP Phone:

Cisco recommends the use of good quality external devices (headsets) that are shielded (screened) against unwanted radio frequency (RF) and audio frequency (AF) signals.

Depending on the quality of these devices and their proximity to other devices such as mobile phones or two-way radios, some audio noise may still occur. In these cases, Cisco recommends that you take one or more of the following actions:

- Move the external device away from the source of the RF or AF signals.
- Route the external device cables away from the source of the RF or AF signals.
- Use shielded cables for the external device, or use cables with a better shield and connector.
- Shorten the length of the external device cable.
- Apply ferrites or other such devices on the cables for the external device.

Cisco cannot guarantee the performance of the system because Cisco has no control over the quality of external devices, cables, and connectors. The system will perform adequately when suitable devices are attached using good quality cables and connectors.

Caution In European Union countries, use only external headsets that are fully compliant with the EMC Directive [89/336/EC].

Installing the Cisco Unified IP Phone

You must connect the Cisco Unified IP Phone to the network and to a power source before using it. See Figure 3-2, Figure 3-3, and Figure 3-4 for a graphical overview of the procedures that follow.

Note Before you install a phone, even if it is new, upgrade the phone to the current firmware image.

Before using external devices, read the Using External Devices with Your Cisco Unified IP Phone, page 3-5 for safety and performance information.

To install a Cisco Unified IP Phone, perform the tasks described in Table 3-1.

Table 3-1 Installing a Cisco Unified IP Phone

Task	Purpose	Related Topics
1.	Connect the footstand to the back of the phone. See Figure 3-1 and Figure 3-2.	—
2.	Connect the handset to the Handset port.	—
3.	Connect the power supply to the Cisco DC Adapter port (DC48V). (Optional) When connecting phones powered by an external power supply, you must connect the power supply to the phone before connecting the Ethernet cable to the phone. When disconnecting the phone, you must disconnect the Ethernet cable before disconnecting the power supply.	See Providing Power to the Cisco Unified IP Phone 7906G and 7911G, page 2-3.
4.	Connect a Category 3 or 5 straight-through Ethernet cable from the switch to the 10/100 SW port. Each Cisco Unified IP Phone ships with one Ethernet cable in the box.	See Network and Access Ports, page 3-3 for guidelines.
5.	(Cisco Unified IP Phone 7911G only) Connect a Category 3 or 5 straight-through Ethernet cable from another network device, such as a desktop computer, to the 10/100 PC port. (Optional) You can connect another network device later if you do not connect one now.	SeeNetwork and Access Ports, page 3-3 for guidelines.

Figure 3-1 Connecting the Footstand (Cisco Unified IP Phone Model 7906G Shown)

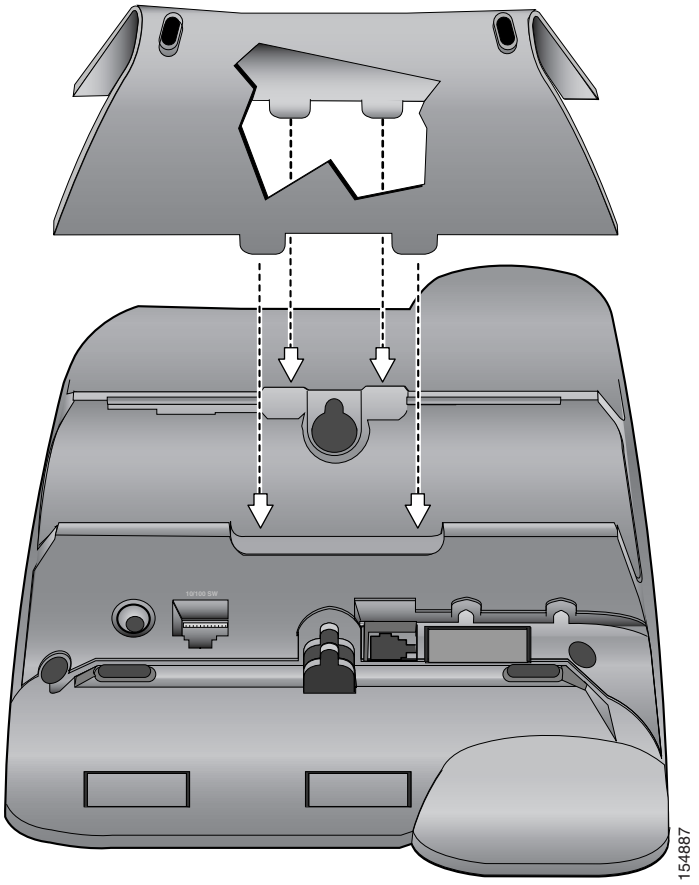


Figure 3-2 Connecting the Footstand (Cisco Unified IP Phone Model 7911G Shown)

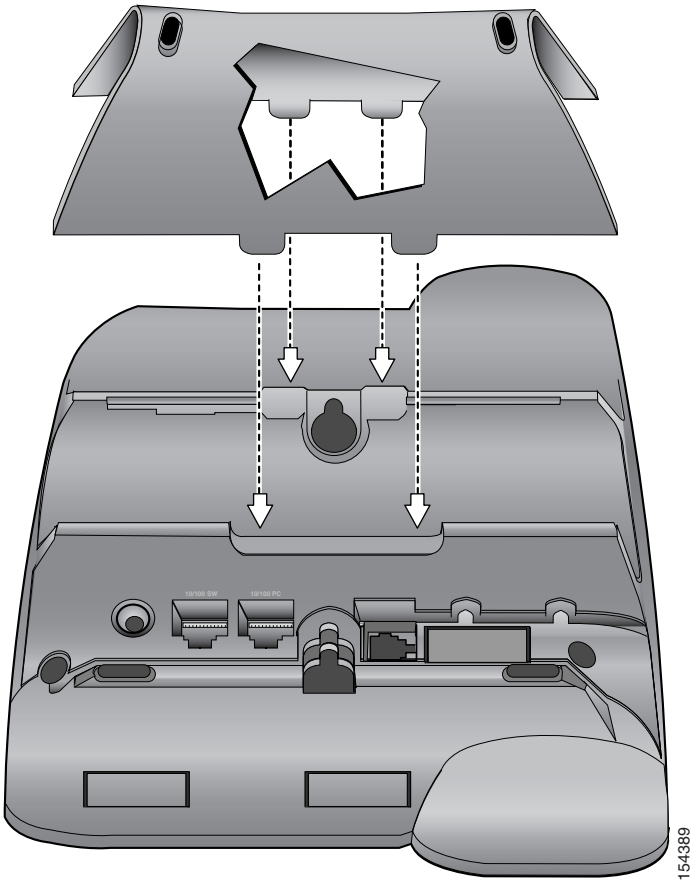
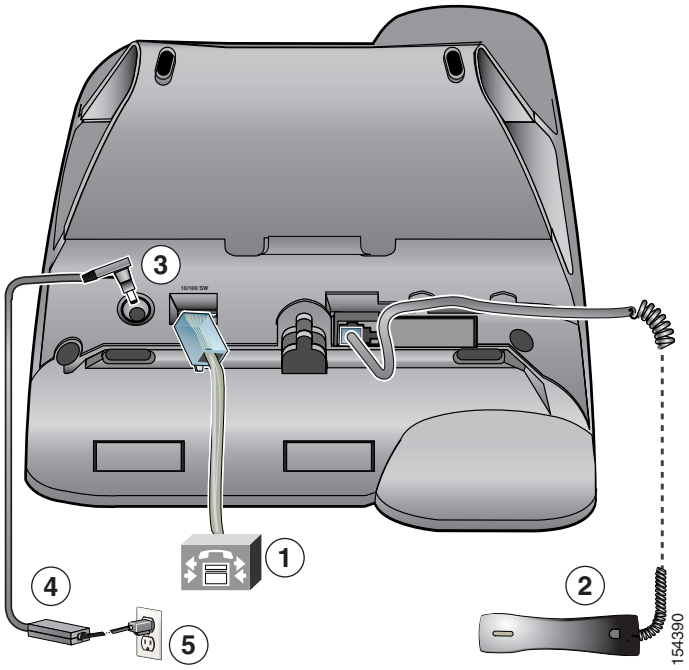
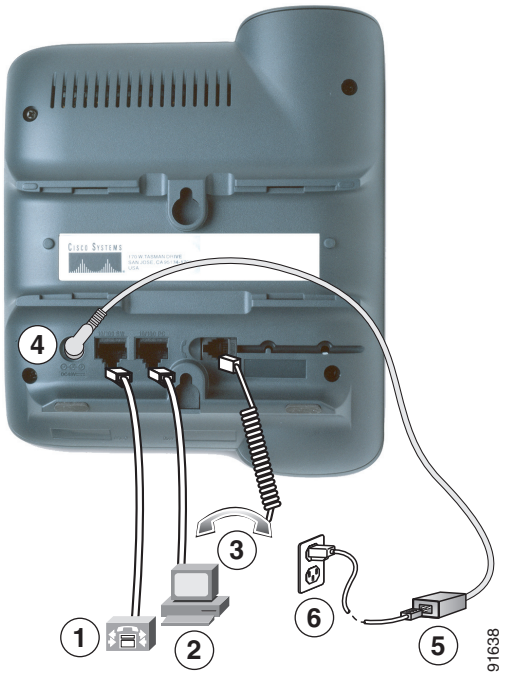


Figure 3-3 Cisco Unified IP Phone Model 7906G Cable Connections



1	Network port (10/100 SW)	**4**	AC-to-DC power supply
2	Handset port	**5**	AC power cord
3	DC Adapter port (DC48V)		

Figure 3-4 Cisco Unified IP Phone Model 7911G Cable Connections



1	Network port (10/100 SW)	**4**	DC Adapter port (DC48V)
2	Access port (10/100 PC)	**5**	AC-to-DC power supply
3	Handset port	**6**	AC power cord

Related Topics

- Before You Begin, page 3-1
- Mounting the Phone to a Wall, page 3-10
- Configuring Startup Network Settings, page 3-11

Mounting the Phone to a Wall

You can mount the Cisco Unified IP Phone on a wall by using the back of the phone as a mounting bracket or you can use special brackets available in a Cisco Unified IP Phone wall mount kit. (Wall mount kits must be ordered separately from the phones.) If you attach the phone to a wall by using the back of the phone and not the wall mount kit, you need to supply the following tools and parts:

- Screwdriver
- Screws to secure the Cisco Unified IP phone to the wall

Before You Begin

To ensure that the handset attaches securely to a wall-mounted phone, remove the handset wall hook from the handset rest, rotate the hook 180 degrees, and reinsert the hook. Turning the hook exposes a lip on which the handset catches when the phone is vertical. For an illustrated procedure, refer to the *Installing the Universal Wall Mount Kit for the Cisco Unified IP Phone* document at: http://www.cisco.com/en/US/products/hw/phones/ps379/prod_installation_guides_list.html

Caution Use care not to damage wires or pipes located inside the wall when securing screws to wall studs.

Procedure

Step 1 Remove the footstand if it is attached to the phone.

Step 2 Insert two screws into a wall stud, matching them to the two screw holes on the back of the phone.

Step 3 Hang the phone on the wall.

Verifying the Phone Startup Process

After the Cisco Unified IP Phone has power connected to it, the phone begins its startup process by cycling through these steps.

1. These buttons blink or flash on and off:
   - Handset light strip
   - Hold button
   - Applications Menu button
2. The screen displays the Cisco Systems, Inc., logo screen.
3. These messages display as the phone starts:
   - Configuring IP
   - Updating Trust List
   - Verifying Load
   - Configuring CM List
   - Registering
4. The main screen displays:
   - Current date and time
   - Directory number
   - Softkeys

If the phone successfully passes through these stages, it has started up properly. If the phone does not start up properly, see Resolving Startup Problems, page 9-1.

Configuring Startup Network Settings

If you are not using DHCP in your network, you must configure these network settings on the Cisco Unified IP Phone after installing the phone on the network:

- IP address
- IP subnet information
- Default gateway IP address

- Domain name
- DNS server IP address
- TFTP server IP address

Collect this information and see the instructions in Configuring Settings on the Cisco Unified IP Phone, page 4-1.

Configuring Security on the Cisco Unified IP Phone

The security features protect against several threats, including threats to the identity of the phone and to data. These features establish and maintain authenticated communication streams between the phone and the Cisco Unified Communications Manager server, and digitally sign files before they are delivered.

For more information about the security features, see Understanding Security Features for Cisco Unified IP Phones, page 1-10. Also, refer to *Cisco Unified Communications Manager Security Guide*.

You can install an Locally Significant Certificate (LSC) from the Security Configuration menu on the phone. This menu also lets you update or remove an LSC.

Before You Begin

Make sure that the appropriate Cisco Unified Communications Manager and the Certificate Authority Proxy Function (CAPF) security configurations are complete:

- The CTL or ITL file should have a CAPF certificate.
- On Cisco Unified Communications Operating System Administration, verify that the CAPF certificate has been installed
- The CAPF is running and configured.
- The phone should have the correct load file. To verify the image, press the **Applications Menu** button and choose **Settings > Model Information**.

Refer to C*isco Unified Communications Manager Security Guide* for more information.

To manually configure an LSC on the phone, follow these steps:

Procedure

Step 1 Obtain the CAPF authentication code that was set when the CAPF was configured.

Step 2 From the phone, press the **Applications Menu** button and choose **Settings > Security Configuration**.

Note You can control access to the Settings Menu by using the Settings Access field in the Cisco Unified Communications Manager Administration Phone Configuration window. For more information, see *Cisco Unified Communications Manager Administration Guide*.

Step 3 Press ****#** to unlock settings on the Security Configuration menu. (See Unlocking and Locking Options, page 4-3 for information using locking and unlocking options.)

Note If a Settings Menu password has been provisioned, SIP phones present an “Enter password” prompt after you enter **#.

Step 4 Scroll to LSC and press the **Update** softkey.

The phone prompts for an authentication string.

Step 5 Enter the authentication code and press the **Submit** softkey.

The phone begins to install, update, or remove the LSC, depending on how the CAPF was configured. During the procedure, a series of messages displays in the LSC option field in the Security Configuration menu so that you can monitor progress. When the procedure completes successfully, the phone will display Installed or Not Installed.

The LSC install, update, or removal process can take a long time to complete. You can stop the process at any time by pressing the **Stop** softkey from the Security Configuration menu. (Settings must be unlocked before you can press this softkey.)

When the phone successfully completes the installation procedure, it displays “Success.” If the phone displays, “Failure,” the authorization string may be incorrect or the phone may not enabled for upgrading. Refer to error messages generated on the CAPF server and take appropriate actions.

You can verify that an LSC is installed on the phone by pressing the **Applications Menu** button, then choosing **Settings > Model Information**, and ensuring that the LSC setting shows Installed.

Related Topic

Understanding Security Features for Cisco Unified IP Phones, page 1-10.

CHAPTER 4

Configuring Settings on the Cisco Unified IP Phone

The Cisco Unified IP Phone includes many configurable network and device settings that you may need to modify before the phone is functional for your users. You can access these settings, and change many of them, through menus on the phone.

This chapter includes the following topics:

- Configuration Menus on the Cisco Unified IP Phones 7906G and 7911G, page 4-1
- Overview of Options Configurable from a Phone, page 4-4
- Network Configuration Menu, page 4-5
- Device Configuration Menu, page 4-17
- Security Configuration Menu, page 4-33

Configuration Menus on the Cisco Unified IP Phones 7906G and 7911G

The Cisco Unified IP Phone includes the following configuration menus:

- Network Configuration—Provides options for viewing and making a variety of network settings. For more information, see Network Configuration Menu, page 4-5.
- Device Configuration—Provides access to sub-menus from which you can view a variety of non network-related settings. For more information, see Device Configuration Menu, page 4-17.
- Security Configuration—Provides options for displaying and modifying security settings. For more information, see Security Configuration Menu, page 4-33.

Before you can change option settings on the Network Configuration menu, you must unlock options for editing. See Unlocking and Locking Options, page 4-3 for instructions.

For information about the keys you can use to edit or change option settings, see Editing the Values of an Option Setting, page 4-3.

You can control whether a phone user has access to phone settings by using the Settings Access field in the Cisco Unified Communications Manager Administration Phone Configuration Settings window. See *Cisco Unified Communications Manager Administration Guide* for more information.

Related Topics

- Unlocking and Locking Options, page 4-3
- Editing the Values of an Option Setting, page 4-3
- Overview of Options Configurable from a Phone, page 4-4
- Network Configuration Menu, page 4-5
- Device Configuration Menu, page 4-17

Displaying a Configuration Menu

To display a configuration menu, perform the following steps.

Note You can control whether a phone has access to the Settings menu or to options on this menu by using the Settings Access field in the Cisco Unified Communications Manager Administration Phone Configuration window. The Settings Access field accepts these values:

- **Enabled**—Allows access to the Settings menu.
- **Disabled**—Prevents access to the Settings menu.
- **Restricted**—Allows access to the User Preferences menu and allows volume changes to be saved. Prevents access to other options on the Settings menu.

If you cannot access an option on the Settings menu, check the Settings Access field.

Procedure

Step 1 Press the **Applications Menu** button.

Step 2 Choose **Settings**.

Step 3 Perform one of these actions to display the desired menu:

- Use the **Navigation** button to select the desired menu and then press the **Select** softkey.
- Use the keypad on the phone to enter the number that corresponds to the menu.

Step 4 To display a submenu, repeat Step 3.

Step 5 To exit a menu, press the **Exit** softkey.

Related Topics

- Unlocking and Locking Options, page 4-3
- Editing the Values of an Option Setting, page 4-3
- Overview of Options Configurable from a Phone, page 4-4
- Network Configuration Menu, page 4-5
- Device Configuration Menu, page 4-17

Unlocking and Locking Options

Configuration options that can be changed from a phone are locked by default to prevent users from making changes that could affect the operation of a phone. You must unlock these options before you can change them.

When options are inaccessible for modification, a *locked* padlock icon appears on the configuration menus. When options are unlocked and accessible for modification, an *unlocked* padlock icon appears on these menus, as shown next.

To unlock or lock options, press **#. This action either locks or unlocks the options, depending on the previous state.

Note If a Settings Menu password has been provisioned, SIP phones present an "Enter password" prompt after you enter **#.

After you have made your changes, you must lock the options.

Caution Do not press **# to unlock options and then immediately press **# again to lock options. The phone will interpret this sequence as **#**, which will reset the phone. To lock options after unlocking them, wait at least 10 seconds before you press **# again.

Related Topics

- Displaying a Configuration Menu, page 4-2
- Editing the Values of an Option Setting, page 4-3
- Overview of Options Configurable from a Phone, page 4-4
- Network Configuration Menu, page 4-5
- Device Configuration Menu, page 4-17

Editing the Values of an Option Setting

When you edit the value of an option setting, follow these guidelines:

- Use the keys on the keypad to enter numbers and letters.
- To enter letters by using the keypad, use a corresponding number key. Press the key one or more times to display a particular letter. For example, press the **2** key once for "a," twice quickly for "b," and three times quickly for "c." After you pause, the cursor automatically advances to allow you to enter the next letter.
- To enter a period (for example, in an IP address under IPv4 Configuration), press the **.** (period) softkey or press * on the keypad.

- To enter a colon (for example, in an IP address under IPv6 Configuration), press the **:** (colon) softkey or press * on the keypad.
- Press the **<<** softkey if you make a mistake. This softkey deletes the character to the left of the cursor.
- Press the **Cancel** softkey before pressing the **Save** softkey to discard any changes that you have made.

Note The Cisco Unified IP Phone provides several methods you can use to reset or restore option settings, if necessary. For more information, see Resetting or Restoring the Cisco Unified IP Phone, page 9-12.

Related Topics

- Displaying a Configuration Menu, page 4-2
- Unlocking and Locking Options, page 4-3
- Overview of Options Configurable from a Phone, page 4-4
- Network Configuration Menu, page 4-5
- Device Configuration Menu, page 4-17

Overview of Options Configurable from a Phone

The settings that you can change on a phone fall into several categories, as shown in Table 4-1. For a detailed explanation of each setting and instructions for changing them, see Network Configuration Menu, page 4-5.

Note There are several options on the Network Configuration menu and on the Device Configuration Menu that are for display only or that you can configure from Cisco Unified Communications Manager. These options are also described in the Network Configuration Menu, page 4-5 and the or the Device Configuration Menu, page 4-17.

Table 4-1 Configurable Settings

Category	Description	Network Configuration Menu Option
General Network Settings		
VLAN settings	Admin. VLAN ID allows you to change the administrative VLAN used by the phone. PC VLAN allows the phone to interoperate with third-party switches that do not support a voice VLAN.	Admin. VLAN ID PC VLAN (applies to 7911G only)
Port settings	Allow you to set the speed and duplex of the network and access ports.	SW Port Configuration PC Port Configuration (applies to 7911G only)
IPv4 Network Settings		
DHCP settings	Dynamic Host Configuration Protocol (DHCP) automatically assigns IP address to devices when you connect them to the network. Cisco Unified IP Phones enable DHCP by default.	DHCP DHCP Address Released

Table 4-1 Configurable Settings (continued)

Category	Description	Network Configuration Menu Option
IP settings	If you do not use DHCP in your network, you can make IP settings manually.	Domain Name
		IP Address
		Subnet Mask
		Default Router 1-5
		DNS Server 1-5
TFTP settings for TFTP IPv4 servers	If you do not use DHCP to direct the phone to a TFTP server, you must manually assign a TFTP server. You can also assign an alternative TFTP server to use instead of the one assigned by DHCP.	Alternate TFTP
		TFTP Server 1
		TFTP Server 2
IPv6 Network Settings		
DHCP settings	Dynamic Host Configuration Protocol (DHCP) automatically assigns IP address to devices when you connect them to the network. Cisco Unified IP Phones enable DHCP by default.	DHCPv6
		DHCPv6 Address Released
IP settings	If you do not use DHCP in your network, you can make IP settings manually.	Domain Name
		IPv6 Address
		IPv6 Prefix Length
		IPv6 Default Router 1-2
		IPv6 DNS Server 1-2
TFTP settings for TFTP IPv6 servers (SCCP phones only)	If you do not use DHCP to direct the phone to a TFTP server, you must manually assign a TFTP server. You can also assign an alternative TFTP server to use instead of the one assigned by DHCP.	IPv6 Alternate TFTP
		IPv6 TFTP Server 1
		IPv6 TFTP Server 2

Related Topics

- Displaying a Configuration Menu, page 4-2
- Unlocking and Locking Options, page 4-3
- Editing the Values of an Option Setting, page 4-3
- Network Configuration Menu, page 4-5
- Device Configuration Menu, page 4-17

Network Configuration Menu

The Network Configuration menu provides options for viewing and making a variety of network settings. Table 4-2, Table 4-3 and Table 4-4 describe these options and, where applicable, explain how to change them.

For information about how to access the Network Configuration menu, see Displaying a Configuration Menu, page 4-2.

Before you can change an option on this menu, you must unlock options as described in the Unlocking and Locking Options, page 4-3. The **Edit**, **Yes**, or **No** softkeys for changing network configuration options appear only if options are unlocked.

For information about the keys you can use to edit options, see Editing the Values of an Option Setting, page 4-3.

Table 4-2 Network Configuration Menu Options

Option	Description	To Change
IPv4 Configuration	Internet Protocol v4 address menu. In the IPv4 Configuration menu, you can do the following: • Enable or disable the phone to use the IPv4 address that is assign by the DHCPv4 server. • Manually set the IPv4 Address, Subnet Mask, Default Routers, DNSv4 Server, and Alternate TFTP servers for IPv4. For more information on the IPv4 address fields, refer to Table 4-3 on page 4-9.	1. Unlock network configuration options. 2. Scroll to IPv4 Configuration and press the **Select** softkey.
IPv6 Configuration	Internet Protocol v6 address menu. In the IPv6 Configuration menu, you can do the following: • Enable or disable the phone to use the IPv6 address that is assigned by the DHCPv6 server or to use the IPv6 address that it acquires through Stateless Address Autoconfiguration (SLAAC). • Manually set the IPv6 Address, Subnet Prefix Length, DNSv6 Server, and IPv6 TFTP Servers. For more information on the IPv6 address fields, refer to Table 4-4 on page 4-13. For more information on SLAAC, refer to *Deploying IPv6 in Unified Communications Networks with Cisco Unified Communications Manager*.	1. Unlock network configuration options. 2. Scroll to IPv6 Configuration and press the **Select** softkey.
MAC Address	Unique Media Access Control (MAC) address of the phone.	Display only—Cannot configure.
Host Name	Unique host name that the DHCP server assigned to the phone.	Display only—Cannot configure.

Table 4-2 Network Configuration Menu Options (continued)

Option	Description	To Change
Domain Name	Name of the Domain Name System (DNS) domain in which the phone resides. **Note** If the phone receives different domain names from the DHCPv4 and DHCPv6 servers, the domain name from the DHCPv6 takes precedence.	1. Unlock network configuration options. 2. Disable DHCP. If the IP Addressing mode is configured for IPv4 only, set the DHCP option to **No**. If the IP Addressing mode is configured for IPv6 only, set the DHCPv6 option to **No**. If the IP Addressing mode is configured for both IPv4 and IPv6, set both DHCP option and DHCPv6 to **No**. 3. Scroll to the Domain Name option, press the **Edit** softkey, and then enter a new domain name. 4. Press the **Validate** softkey and then press the **Save** softkey.
Operational VLAN ID	Auxiliary Virtual Local Area Network (VLAN) configured on a Cisco Catalyst switch in which the phone is a member. If the phone has not received an auxiliary VLAN, this option indicates the Administrative VLAN. If neither the auxiliary VLAN nor the Administrative VLAN are configured, this option is blank.	The phone obtains its Operational VLAN ID via Cisco Discovery Protocol (CDP) from the switch to which the phone is attached. To assign a VLAN ID manually, use the Admin VLAN ID option.
Admin. VLAN ID	Auxiliary VLAN in which the phone is a member. Used only if the phone does not receive an auxiliary VLAN from the switch; otherwise it is ignored.	1. Unlock network configuration options. 2. Scroll to the Admin. VLAN ID option, press the **Edit** softkey, and then enter a new Admin VLAN setting. 3. Press the **Validate** softkey and then press the **Save** softkey.

Table 4-2 Network Configuration Menu Options (continued)

Option	Description	To Change
SW Port Configuration	Speed and duplex of the network port (labeled 10/100 SW on the Cisco Unified IP Phone 7970, and 10/100/1000 SW on the Cisco Unified IP Phone 7971G-GE). Valid values: • Auto Negotiate • 10 Half—10-BaseT/half duplex • 10 Full—10-BaseT/full duplex • 100 Half—100-BaseT/half duplex • 100 Full—100-BaseT/full duplex • 1000 Full—1000-BaseT/full duplex If the phone is connected to a switch, configure the port on the switch to the same speed/duplex as the phone, or configure both to auto-negotiate. If you change the setting of this option, you must change the PC Port Configuration option to the same setting.	1. Unlock network configuration options. 2. Scroll to the SW Port Configuration option and then press the **Edit** softkey. 3. Scroll to the setting that you want and then press the **Select** softkey. 4. Press the **Save** softkey.
PC Port Configuration	Speed and duplex of the access port (labeled 10/100 PC on the Cisco Unified IP Phone 7970, and 10/100/1000 PC on the Cisco Unified IP Phone 7971G-GE). Valid values: • Auto Negotiate • 10 Half—10-BaseT/half duplex • 10 Full—10-BaseT/full duplex • 100 Half—100-BaseT/half duplex • 100 Full—100-BaseT/full duplex • 1000 Full—1000-BaseT/full duplex If the phone is connected to a switch, configure the port on the switch to the same speed/duplex as the phone, or configure both to auto-negotiate. If you change the setting of this option, you must change the SW Port Configuration option to the same setting.	1. Unlock network configuration options. 2. Scroll to the PC Port Configuration option and then press the **Edit** softkey. 3. Scroll to the setting that you want and then press the **Select** softkey. 4. Press the **Save** softkey.
PC VLAN	Allows the phone to interoperate with 3rd party switches that do not support a voice VLAN. The Admin VLAN ID option must be set before you can change this option.	1. Unlock network configuration options. 2. Make sure the Admin VLAN ID option is set. 3. Scroll to the **PC VLAN** option, press the **Edit** softkey, and then enter a new PC VLAN setting. 4. Press the **Validate** softkey and then press the **Save** softkey.

Table 4-3 describes the IPv4 configuration menu options.

Table 4-3 IPv4 Configuration Menu Options

Option	Description	To Change
DHCP	Indicates whether the phone has DHCP enabled or disabled. When DHCP is enabled, the DHCP server assigns the phone an IPv4 address. When DHCP is disabled, the administrator must manually assign an IPv4 address to the phone.	1. Unlock network configuration options. 2. Scroll to the DHCP option and press the **No** softkey to disable DHCP, or press the **Yes** softkey to enable DHCP. 3. Press the **Save** softkey.
IP Address	Internet Protocol version 4 (IPv4) address of the phone. If you assign an IPv4 address with this option, you must also assign a subnet mask and default router. See the Subnet Mask and Default Router options in this table.	1. Unlock network configuration options. 2. Set the DHCP option to **No**. 3. Scroll to the IP Address option, press the **Edit** softkey, and then enter a new IP Address. 4. Press the **Validate** softkey and then press the **Save** softkey.
Subnet Mask	Subnet mask used by the phone.	1. Unlock network configuration options. 2. Set the DHCP option to No. 3. Scroll to the Subnet Mask option, press the **Edit** softkey, and then enter a new subnet mask. 4. Press the **Validate** softkey and then press the **Save** softkey.
Default Router 1 Default Router 2 Default Router 3 Default Router 4 Default Router 5	Default router used by the phone (Default Router 1) and optional backup routers (Default Router 2–5).	1. Unlock network configuration options. 2. Set the DHCP option to No. 3. Scroll to the appropriate Default Router option, press the **Edit** softkey, and then enter a new router IP address. 4. Press the **Validate** softkey. 5. Repeat Steps 3 and 4 as needed to assign backup routers. 6. Press the **Save** softkey.
DNS Server 1 DNS Server 2 DNS Server 3 DNS Server 4 DNS Server 5	Primary Domain Name System (DNS) server (DNS Server 1) and optional backup DNS servers (DNS Server 2–5) used by the phone.	1. Unlock network configuration options. 2. Set the DHCP option to No. 3. Scroll to the appropriate DNS Server option, press the **Edit** softkey, and then enter a new DNS server IP address. 4. Press the **Validate** softkey. 5. Repeat Steps 3 and 4 as needed to assign backup DNS servers. 6. Press the **Save** softkey.

Table 4-3 IPv4 Configuration Menu Options (continued)

Option	Description	To Change
DHCP Server	IP address of the Dynamic Host Configuration Protocol (DHCP) server from which the phone obtains its IPv4 address.	Display only—Cannot configure.
DHCP Address Released	Releases the IPv4 address assigned by DHCP.	1. Unlock network configuration options. 2. Scroll to the DHCP Address Released option and press the **Yes** softkey to release the IP address assigned by DHCP, or press the **No** softkey if you do not want to release this IP address. 3. Press the **Save** softkey.
Alternate TFTP	Indicates whether the phone is using an alternative TFTP server.	1. Unlock network configuration options. 2. Scroll to the Alternate TFTP option and press the **Yes** softkey if the phone should use an alternative TFTP server. 3. Press the **Save** softkey.

Table 4-3 IPv4 Configuration Menu Options (continued)

Option	Description	To Change
TFTP Server 1	Primary Trivial File Transfer Protocol (TFTP) server used by the phone. If you are not using DHCP in your network and you want to change this server, you must use the TFTP Server 1 option. If you set the Alternate TFTP option to yes, you must enter a non-zero value for the TFTP Server 1 option. If neither the primary TFTP server nor the backup TFTP server is listed in the CTL or ITL file on the phone, you must unlock either of the files before you can save changes to the TFTP Server 1 option. In this case, the phone will delete either of the files when you save changes to the TFTP Server 1 option. A new CTL or ITL file will be downloaded from the new TFTP Server 1 address. When the phone looks for its TFTP server, the phone gives precedence to manually assigned TFTP servers, regardless of the protocol. If your configuration includes both IPv6 and IPv4 TFTP servers, the phone prioritizes the order that it looks for its TFTP server by giving priority to manually assigned IPv6 TFTP servers and IPv4 TFTP servers. The phone looks for its TFTP server in the following order: 1. Manually assigned IPv6 TFTP Servers 2. Manually assigned IPv4 TFTP Servers 3. DHCPv6 assigned TFTP Servers 4. DHCP assigned TFTP Servers **Note** For information about the CTL and ITL files, refer to *Cisco Unified Communications Manager Security Guide*. For information about unlocking the CTL and ITL files, see Unlocking the CTL and ITL files, page 4-35.	1. Unlock the CTL or ITL file if necessary (for example, if you are changing the administrative domain of the phone). If both the CTL and ITL files exist, unlock either of the files. 2. If DHCP is enabled, set the Alternate TFTP option to Yes. 3. Scroll to the TFTP Server 1 option, press the **Edit** softkey, and then enter a new TFTP server IP address. 4. Press the **Validate** softkey, and then press the **Save** softkey.

Table 4-3 IPv4 Configuration Menu Options (continued)

Option	Description	To Change
TFTP Server 2	Optional backup TFTP server that the phone uses if the primary TFTP server is unavailable. If neither the primary TFTP server nor the backup TFTP server is listed in the CTL or ITL file on the phone, you must unlock either of the files before you can save changes to the TFTP Server 2 option. In this case, the phone will delete either of the files when you save changes to the TFTP Server 2 option. A new CTL or ITL file will be downloaded from the new TFTP Server 2 address. When the phone looks for its TFTP server, the phone gives precedence to manually assigned TFTP servers, regardless of the protocol. If your configuration includes both IPv6 and IPv4 TFTP servers, the phone prioritizes the order that it looks for its TFTP server by giving priority to manually assigned IPv6 TFTP servers and IPv4 TFTP servers. The phone looks for its TFTP server in the following order: 1. Manually assigned IPv6 TFTP Servers 2. Manually assigned IPv4 TFTP Servers 3. DHCPv6 assigned TFTP Servers 4. DHCP assigned TFTP Servers For information about the CTL or ITL file, refer to *Cisco Unified Communications Manager Security Guide*. For information about unlocking the CTL and ITL files, see Unlocking the CTL and ITL files, page 4-35.	1. Unlock the CTL or ITL file if necessary (for example, if you are changing the administrative domain of the phone). If both the CTL and ITL files exist, unlock either of the files. 2. Unlock network configuration options. 3. Enter an IP address for the TFTP Server 1 option. 4. Scroll to the TFTP Server 2 option, press the **Edit** softkey, and then enter a new backup TFTP server IP address. 5. Press the **Validate** softkey, and then press the **Save** softkey. **Note** If you forgot to unlock the CTL or ITL file, you can change the TFTP Server 2 address in either file, then erase them by pressing the Erase softkey from the Security Configuration menu. A new CTL or ITL file will be downloaded from the new TFTP Server 2 address.
BOOTP Server	Indicates whether the phone obtains its configuration from a Bootstrap Protocol (BootP) server instead of from a DHCP server.	Display only—Cannot configure.

Table 4-4 describes the IPv6 configuration menu options.

Table 4-4 IPv6 Configuration Menu Options

Option	Description	To Change
DHCPv6	Indicates whether the phone has DHCP enabled or disabled. When DHCPv6 is enabled, the DHCPv6 server assigns the phone an IPv6 address. When DHCP v6 is disabled, the administrator must manually assign an IPv6 address to the phone. The DHCPv6 setting along with the Auto IP Configuration setting determine how the IP Phone obtains its network settings. For more information on how these two settings affect the network settings on the phone, see Table 4-5.	1. Unlock network configuration options. 2. Scroll to the DHCPv6 option and press the **No** softkey to disable DHCP, or press the **Yes** softkey to enable DHCP. 3. Press the **Save** softkey.
IPv6 Address	Internet Protocol (IP) address of the phone. The IPv6 address is a 128 bit address. If you assign an IP address with this option, you must also assign the IPv6 prefix length and default router. See the IPv6 Subnet Prefix option in this table.	1. Unlock network configuration options. 2. Set the DHCPv6 option to **No**. 3. Scroll to the IP Address option, press the **Edit** softkey, and then enter a new IP Address. 4. Press the **Validate** softkey and then press the **Save** softkey.
IPv6 Prefix Length	Subnet prefix length that is us ed by the phone. The subnet prefix length is a decimal value from 1-128, that specifies the portion of the IPv6 address that comprises the subnet.	1. Unlock network configuration options. 2. Set the DHCPv6 option to **No**. 3. Scroll to the IPv6 Prefix Length option, press the **Edit** softkey, and then enter a new subnet mask. 4. Press the **Validate** softkey and then press the **Save** softkey.
IPv6 Default Router 1	Default router used by the phone (Default Router 1). **Note** The phone obtains information on the default router from IPv6 Router Advertisements.	1. Unlock network configuration options. 2. Set the DHCPv6 option to **No**. 3. Scroll to the appropriate Default Router option, press the **Edit** softkey, and then enter a new router IP address. 4. Press the **Validate** softkey. 5. Repeat Steps 3 and 4 as needed to assign the backup router. 6. Press the **Save** softkey.

Table 4-4 IPv6 Configuration Menu Options (continued)

Option	Description	To Change
IPv6 DNS Server 1 IPv6 DNS Server 2	Primary Domain Name System (DNS) server (DNS Server 1) and optional backup DNS servers (DNS Server 2) used by the phone. If your configuration includes both DNSv6 and DNSv4 servers, the phone looks for its DNS server in the following order: 1. IPv6 DNS Server 1 2. IPv6 DNS Server 2 3. DNS Server 1-5 for IPv4 (respectively)	1. Unlock network configuration options. 2. Set the DHCPv6 option to **No**. 3. Scroll to the appropriate DNS Server option, press the **Edit** softkey, and then enter a new DNS server IP address. 4. Press the **Validate** softkey. 5. Repeat Steps 3 and 4 as needed to assign the backup DNS server. 6. Press the **Save** softkey.
DHCPv6 Address Released	Releases the IPv6 address that is assigned by DHCP.	1. Unlock network configuration options. 2. Scroll to the DHCPv6 Address Released option and press the **Yes** softkey to release the IP address assigned by DHCP, or press the **No** softkey if you do not want to release this IP address. 3. Press the **Save** softkey.
IPv6 Alternate TFTP	Indicates whether the phone is using an alternative TFTP server.	1. Unlock network configuration options. 2. Scroll to the IPv6 Alternate TFTP option and press the **Yes** softkey if the phone should use an alternative TFTP server. 3. Press the **Save** softkey.

Table 4-4 IPv6 Configuration Menu Options (continued)

Option	Description	To Change
IPv6 TFTP Server 1 (SCCP phones only)	Primary IPv6 Trivial File Transfer Protocol (TFTP) server used by the phone. If you are not using DHCPv6 in your network and you want to change this server, you must use the IPv6 TFTP Server 1 option. If you set the IPv6 Alternate TFTP option to Yes or you disable DHCPv6, you must enter a non-zero value for the IPv6 TFTP Server 1 option. **Note** Before making changes to the Alternate TFTP or IPv6 TFTP servers, you must first unlock the CTL or ITL file on the phone. When the phone looks for its TFTP server, it gives precedence to manually assigned TFTP servers, regardless of the protocol. If your configuration includes both IPv6 and IPv4 TFTP servers, the phone prioritizes the order that it looks for its TFTP server by giving priority to manually assigned IPv6 TFTP servers and IPv4 TFTP servers. The phone looks for its TFTP server in the following order: 1. Manually assigned IPv6 TFTP Servers 2. Manually assigned IPv4 TFTP Servers 3. DHCPv6 assigned TFTP Servers 4. DHCP assigned TFTP Servers For information about the CTL or ITL file, refer to *Cisco Unified Communications Manager Security Guide*. For information about unlocking the CTL or ITL file, see Unlocking the CTL and ITL files, page 4-35.	1. Unlock the CTL or ITL file if necessary. If both the CTL and ITL files exist, unlock either of the files. 2. If DHCPv6 is enabled, set the Alternate TFTP option to **Yes**. 3. Scroll to the IPv6 TFTP Server 1 option, press the **Edit** softkey, and then enter a new TFTP server IP address. 4. Press the **Validate** softkey, and then press the **Save** softkey.

Table 4-4 IPv6 Configuration Menu Options (continued)

Option	Description	To Change
IPv6 TFTP Server 2 (for SCCP phones only)	(Optional) The backup IPv6 TFTP server that the phone uses if the primary IPv6 TFTP server is unavailable. **Note** Before making changes to the Alternate TFTP or IPv6 TFTP servers, you must first unlock the CTL or ITL file on the phone. When the phone looks for its TFTP server, it gives precedence to manually assigned TFTP servers, regardless of the protocol. If your configuration includes both IPv6 and IPv4 TFTP servers, the phone prioritizes the order that it looks for its TFTP server by giving priority to manually assigned IPv6 TFTP servers and IPv4 TFTP servers. The phone looks for its TFTP server in the following order: 1. Manually assigned IPv6 TFTP Servers 2. Manually assigned IPv4 TFTP Servers 3. DHCPv6 assigned TFTP Servers 4. DHCP assigned TFTP Servers For information about the CTL or ITL file, refer to *Cisco Unified Communications Manager Security Guide*. For information about unlocking the CTL or ITL file, see Unlocking the CTL and ITL files, page 4-35.	1. Unlock the CTL or ITL file if necessary. If both the CTL and ITL files exist, unlock either of the files. 2. Unlock network configuration options. 3. Enter an IP address for the IPv6 TFTP Server 1 option. 4. Scroll to the IPv6 TFTP Server 2 option, press the **Edit** softkey, and then enter a new backup TFTP server IP address. 5. Press the **Validate** softkey, and then press the **Save** softkey.

Understanding DHCPv6 and Autoconfiguration

You can choose to configure the IP address and other network settings, such as the TFTP server, DNS server, domain, name, and so forth. on an IP phone manually or by using a router and/or a DHCP server to automatically assign the IP address and other network information. For more information on how the Auto IP Configuration and DHCPv6 settings determine where the IP Phone acquires its IPv6 address and other network settings, see Table 4-5.

Table 4-5 Determining How a Phone Acquires its Network Settings

DHCPv6	Allow AutoConfiguration for Phones	How the Phone Acquires its IP address and Network Settings
Disabled	Disabled	You must manually configure an IP address and other network settings. **Note** When DHCPv6 is disabled, the Auto IP Configuration setting is ignored.
Disabled	Enabled	You must manually configure an IP address and the other network settings. **Note** When DHCPv6 is disabled, the Auto IP Configuration setting is ignored.

Table 4-5 Determining How a Phone Acquires its Network Settings (continued)

DHCPv6	Allow AutoConfiguration for Phones	How the Phone Acquires its IP address and Network Settings
Enabled	Disabled	The DHCP server assigns the IP address and the other network settings to the phone.
Enabled	Enabled	When the M-bit is set on the router, the O-bit is ignored. The phone can set its IPv6 address based on an IPv6 address that it received from a DHCPv6 server or the phone can acquire its IPv6 address through stateless address autoconfiguration. When the M-bit is not set, you should set the O-bit on the router. The phone will then acquire its IPv6 address through stateless address autoconfiguration. It will not request an IPv6 address from the DHCPv6 server, but it will request other network configuration information.

Related Topics

- Displaying a Configuration Menu, page 4-2
- Unlocking and Locking Options, page 4-3
- Editing the Values of an Option Setting, page 4-3
- Overview of Options Configurable from a Phone, page 4-4
- Device Configuration Menu, page 4-17

Device Configuration Menu

The Device Configuration menu provides access to submenus from which you can view a variety of settings that are specified in the configuration file for a phone. (The phone downloads the configuration file from the TFTP server.) These sub-menus are:

- Unified CM Configuration Menu, page 4-18
- SIP Configuration Menu for SIP Phones, page 4-19
- Call Preferences Menu for SIP Phones, page 4-21
- HTTP Configuration Menu, page 4-22
- Locale Configuration Menu, page 4-23
- UI Configuration Menu, page 4-23
- Media Configuration Menu, page 4-25
- NTP Configuration Menu for SIP Phones, page 4-26
- Ethernet Configuration Menu, page 4-26
- Security Configuration Menu, page 4-27
- Security Configuration Menu, page 4-27
- QoS Configuration Menu, page 4-28
- Network Configuration, page 4-28

For instructions about how to access the Device Configuration menu and its sub-menus, see Displaying a Configuration Menu, page 4-2.

Unified CM Configuration Menu

The Unified CM Configuration menu contains the options Unified CM1, Unified CM2, Unified CM3, Unified CM4, and Unified CM5. These options show Cisco Unified Communications Manager servers that are available for processing calls from the phone, in prioritized order. To change these options, use Cisco Unified Communications Manager Administration.

For an available Cisco Unified Communications Manager server, an option on the Unified CM Configuration menu will show the Cisco Unified Communications Manager server IP address or name and one of the states shown in Table 4-6.

Table 4-6 Cisco Unified Communications Manager Server States

State	Description
Active	Cisco Unified Communications Manager server from which the phone is currently receiving call-processing services
Standby	Cisco Unified Communications Manager server to which the phone switches if the current server becomes unavailable
Blank	No current connection to this Cisco Unified Communications Manager server

An option may also display one of more of the designations or icons shown in Table 4-7.

Table 4-7 Cisco Unified Communications Manager Server Designations

Designation	Description
SRST	Indicates a Survivable Remote Site Telephony router capable of providing Cisco Unified Communications Manager functionality with a limited feature set. This router assumes control of call processing if all other Cisco Unified Communications Manager servers become unreachable. The SRST Cisco Unified Communications Manager always appears last in the list of servers, even if it is active. For more information, refer to Survivable Remote Site Telephony Configuration in the *Cisco Unified Communications Manager Administration Guide*.
TFTP	Indicates that the phone was unable to register with a Cisco Unified Communications Manager listed in its configuration file, and that it registered with the TFTP server instead.
 (Authentication icon)	Indicates that the call is from a trusted device, and that the connection to the Cisco Unified Communications Manager is authenticated. For more information about authentication, refer to *Cisco Unified Communications Manager Security Guide*.
 (Encryption icon)	Indicates that the call is from a trusted device, and that the connection to the Cisco Unified Communications Manager is authenticated and encrypted. For more information about authentication and encryption, refer to *Cisco Unified Communications Manager Security Guide*. The Encryption icon is also displayed when a Cisco Unified IP phone is configured as *protected*. For more information about protected calls, refer to *Cisco Unified Communications Manager Security Guide*. Protected calls are not authenticated.

SIP Configuration Menu for SIP Phones

The SIP Configuration menu is available on SIP phones. This menu contains the following sub-menus:

- SIP General Configuration Menu, page 4-19
- Line Settings Menu for SIP Phones, page 4-20

SIP General Configuration Menu

The SIP General Configuration menu displays information about the configurable SIP parameters on the phone. Table 4-8 describes the options in this menu.

Table 4-8 SIP General Configuration Menu Options

Option	Description	To Change
Preferred CODEC	Displays the CODEC to use when a call is initiated.	Display only—cannot configure.
Out of Band DTMF	Displays the configuration of the out-of-band signaling (for tone detection on the IP side of a gateway). The Cisco Unified IP Phone (SIP) supports out-of-band signaling by using the AVT tone method. Valid values are none, avt, and avt_always.	Display only—cannot configure.
Register with Proxy	This value is set to Yes.	Display only—cannot configure.
Register Expires	Displays the amount of time, in seconds, after which a registration request expires.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > SIP Profile**.
Phone Label	Displays the text that is displayed on the top right status line of the display on the phone. This text is for end-user display only and has no effect on caller identification or messaging.	Display only—cannot configure.
Enable VAD	The default value is set to No.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > SIP Profile**.
VOIP Control Port	Displays the UDP ports used for SIP messages.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > SIP Profile**.
Start Media Port	Displays the start Real-Time Transport Protocol (RTP) range for media.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > SIP Profile**.
End Media Port	Displays the end Real-Time Transport Protocol (RTP) range for media.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > SIP Profile**.
NAT Enabled	Displays if Network Address Translation (NAT) is enabled.	Display only—cannot configure.

Table 4-8 SIP General Configuration Menu Options (continued)

Option	Description	To Change
NAT Address	Displays the WAN IP address of the NAT or firewall server.	Display only—cannot configure.
Call Statistics	The default value is set to No.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > SIP Profile**.

Related Topics

- Displaying a Configuration Menu, page 4-2
- Device Configuration Menu, page 4-17

Line Settings Menu for SIP Phones

The Line Settings menu displays information that relates to the configurable parameters for each of the lines on your SIP phone. Table 4-9 describes the options in this menu.

Table 4-9 Line Settings Menu Options

Option	Description	To Change
Name	Displays the lines and the number used to register each line.	Use Cisco Unified Communications Manager Administration to modify.
Short Name	Displays the short name configured for the line.	Use Cisco Unified Communications Manager Administration to modify.
Longer Authentication Name	Displays the name used by the phone for authentication if a registration is challenged by the proxy server during initialization. The length of the SIP digest authentication name has been increased to 128 characters for Cisco Unified 7900 Series SIP phones. The authentication name is used to verify that the phone is allowed to send SIP messages (REGISTER, INVITE, and SUBSCRIBE) to the Cisco Unified Communications Manager.	Use Cisco Unified Communications Manager Administration to modify.
Authentication Password	Displays the password used by the phone for authentication if a registration is challenged by the proxy server during initialization.	Use Cisco Unified Communications Manager Administration to modify.
Display Name	Displays the identification the phone uses for display for caller identification purposes.	Use Cisco Unified Communications Manager Administration to modify.

Table 4-9 Line Settings Menu Options (continued)

Option	Description	To Change
Proxy Address	Displays the IP address of the proxy server that will be used by the phone. The value is left blank because it is not applicable to SIP phones that are using Cisco Unified Communications Manager.	Display only—cannot configure.
Proxy Port	The value is left blank because it is not applicable to SIP phones that are using Cisco Unified Communications Manager.	Display only—cannot configure.
Shared Line	Displays if the line is part of a shared line (Yes) or not (No).	Display only—cannot configure.

Related Topics

- Displaying a Configuration Menu, page 4-2
- Device Configuration Menu, page 4-17

Call Preferences Menu for SIP Phones

The Call Preferences menu displays settings that relate to the settings for the call preferences on a SIP phone. Table 4-10 describes the options in this menu.

Table 4-10 Call Preferences Menu Options

Option	Description	To Change
Caller ID Blocking	Indicates whether caller ID blocking is enabled (Yes) or disabled (No) for the phone.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > SIP Profile**.
Anonymous Call Block	Indicates whether anonymous call block is enabled (Yes) or disabled (No) for the phone.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > SIP Profile**.
Call Waiting	Indicates whether call waiting is enabled (Yes) or disabled (No) for the phone.	From Cisco Unified Communications Manager Administration, choose **Call Routing > Directory Number**.
Call Hold Ringback	Indicates whether the call hold ringback feature is enabled (Yes) or disabled (No) for the phone.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > SIP Profile**.
Shutter Msg Waiting	Indicates whether shutter message waiting is enabled (Yes) or disabled (No) for the phone.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > SIP Profile**.
Auto Answer Preferences	Indicates whether auto answer is enabled (Yes) or disabled (No) for the phone.	From Cisco Unified Communications Manager Administration, choose **Call Routing > Directory Number**.
Speed Dials	Indicates whether speed dial is enabled (Yes) or disabled (No) for the phone.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Add a New Speed Dial**.

HTTP Configuration Menu

The HTTP Configuration menu displays the URLs of servers from which the phone obtains a variety of information. This menu also displays information about the idle display on the phone.

Note Cisco Unified IP Phones do not support URLs with IPv6 addresses in the URL. This includes hostname which maps to a IPv6 address for directories, services, messages, and information URLs. If you support the phone's usage of URLs, you must configure the phone and the servers that provide URL services with IPv4 addresses.

Table 4-11 describes the HTTP Configuration menu options.

Table 4-11 HTTP Configuration Menu Options

Option	Description	To Change
Directories URL	URL of the server from which the phone obtains directory information.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Services URL	URL of the server from which the phone obtains Cisco Unified IP Phone services.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Messages URL	URL of the server from which the phone obtains message services.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Information URL	URL of the help text that appears on the phone.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Authentication URL	URL that the phone uses to validate requests made to the phone web server.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Proxy Server URL	URL of proxy server, which makes HTTP requests to non-local host addresses on behalf of the phone HTTP client and provides responses from the non-local host to the phone HTTP client.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Idle URL	URL of an XML service that the phone displays when the phone has not been used for the time specified in the Idle URL Time option and no menu is open. For example, you could use the Idle URL option and the Idle URL Timer option to display a stock quote or a calendar on the LCD screen when the phone has not been used for 5 minutes.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Idle URL Time	Number of seconds that the phone has not been used and no menu is open before the XML service specified in the Idle URL option is activated.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.

Locale Configuration Menu

The Locale Configuration menu displays information about the user locale and the network locale used by the phone. Table 4-12 describes the options on this menu.

Table 4-12 Locale Configuration Menu Options

Option	Description	To Change
User Locale	User locale associated with the phone user. The user locale identifies a set of detailed information to support users, including language, font, date and time formatting, and alphanumeric keyboard text information. For more information on installing user locale, refer to the *Cisco Unified Communications Operating System Administration Guide*.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
User Locale Version	Version of the user locale loaded on the phone.	Display only—cannot configure.
User Locale Char Set	Character set that the phone uses for the user locale.	Display only—cannot configure.
Network Locale	Network locale associated with the phone user. The network locale identifies a set of detailed information that supports the phone in a specific location, including definitions of the tones and cadences used by the phone.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Network Locale Version	Version of the network locale loaded on the phone.	Display only—cannot configure.
Time Format 24hr	Displays the time format that is configured on the phone. If set to Yes, the phone will display the time in 24 hour format. If set to No, the phone will display time in 12 hour format.	Use Cisco Unified Communications Manager Administration to modify.
NTP Configuration (SIP phones only)	Menu to view information on NTP server and mode configuration. For more information, see Network Configuration Menu, page 4-5.	From Cisco Unified Communications Manager Administration, choose **System > Phone NTP Reference**.

UI Configuration Menu

The UI Configuration menu displays whether the group listen function is enabled. Use Cisco Unified Communications Manager Administration to modify.

Table 4-13 UI Configuration Menu Options

Option	Description	To Change
Group Listen, Enabled/Disabled	Indicates whether the group listen feature is enabled or disabled.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Reverting Focus Priority	Indicates whether the phone shifts the call focus on the phone screen to an incoming call or a reverting hold call. Settings include: **Lower**—Focus priority given to incoming calls. **Higher** —Focus priority given to reverting calls. **Even** —Focus priority given to the first call.	Use Cisco Unified Communications Manager to modify options. See also: Hold Reversion.
Auto Call Select	Indicates whether the phone automatically shifts the call focus to an incoming call on the same line when the user is already on a call. When this option is enabled, the phone shifts the call focus to the most recent incoming call. When this option is disabled, all automatic focus changes are disabled regardless of their settings. Default: Enabled.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
"more" Softkey Timer	Indicates the number of seconds that additional softkeys are displayed after the user presses **more**. If this timer expires before the user presses another softkey, the display reverts to the initial softkeys. Range: 5 to 30; 0 represents an infinite timer. Default: 5.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Wideband Handset UI Control	Indicates whether the user can configure the Wideband Handset option in the phone user interface. Values: • Enabled—The user can configure the Wideband Handset option in the Audio Preferences menu on the phone (choose **> User Preferences > Audio Preferences> Wideband Handset**). • Disabled—The value of the Wideband Handset option in Cisco Unified Communications Manager Administration gets used (see Media Configuration Menu, page 4-25). Default: Enabled	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.

Table 4-13 UI Configuration Menu Options (continued)

Option	Description	To Change
Personalization	Indicates whether the phone has been enabled for configuring custom ring tones and wallpaper images.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Enbloc Dialing (SCCP only)	Indicates whether the phone will use Enbloc dialing. If "Enabled", the phone will use Enbloc dialing when possible. If "Disabled", the phone will not use Enbloc dialing. You should disable Enbloc dialing if either Forced Authorization Codes (FAC) or Client Matter Codes (CMC) dialing is being used. Default: Enabled	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.

Media Configuration Menu

The Media Configuration menu displays whether the speaker capability is enabled. Table 4-14 describes the options on this menu.

Table 4-14 Media Configuration Menu Options

Option	Description	To Change
Speaker Enabled	Indicates whether the speaker is enabled for monitoring calls on the phone.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Wideband Handset	Indicates whether wideband is enabled or disabled for the handset. Default: "Use Phone Default" on Cisco Unified Communications Manager Administration. (This default means that the phone will be enabled for a wideband handset only if the phone was shipped with a wideband handset.)	• If Wideband Handset UI Control is enabled, you or the user can choose > **User Preferences > Audio Preferences > Wideband Handset.** • If Wideband Handset UI Control is disabled, from Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration** to set this value. **Note** If you allowed this option to be user controllable (in the Wideband Handset UI Control option), the user-configured value takes precedence.

Table 4-14 Media Configuration Menu Options (continued)

Option	Description	To Change
Enterprise Advertise G.722 Codec	Enables/disables Cisco Unified IP Phones to advertise the G.722 codec to Cisco Unified Communications Manager. For more information, see *Cisco Unified Communications Manager System Guide*, Cisco Unified IP Phones. **Note** When a phone is registered with a Cisco Unified Communications Manager that does not support this setting, the default is "Disabled."	From Cisco Unified Communications Manager Administration, choose **System > Enterprise Parameters**.
Device Advertise G.722 Codec	Allows you to override the Enterprise Advertise G.722 Codec on a per-phone basis. The default is "Use System Default," which means the value configured for the Enterprise Advertise G.722 Codec parameter gets used.	• From Cisco Unified Communications Manager Administration, choose **Device > Phone**.

NTP Configuration Menu for SIP Phones

The NTP Configuration menu, which opens when you select NTP Configuration on the Locale Configuration menu, displays information about the NTP server and mode configuration used by the phone. Table 4-15 describes the options on this menu. For more information, see Locale Configuration Menu, page 4-23.

Table 4-15 NTP Configuration Menu Options

Option	Description	To Change
NTP IP Address 1	The IP address of the primary NTP server.	From Cisco Unified Communications Manager Administration, choose **System > Phone NTP Reference.**
NTP IP Address 2	The IP address of the secondary or backup NTP server.	From Cisco Unified Communications Manager Administration, choose **System > Phone NTP Reference.**
NTP Mode 1	The primary server mode. Supported modes are Directed Broadcast and Unicast.	From Cisco Unified Communications Manager Administration, choose **System > Phone NTP Reference.**
NTP Mode 2	The secondary server mode. Supported modes are Directed Broadcast and Unicast.	From Cisco Unified Communications Manager Administration, choose **System > Phone NTP Reference.**
DST Auto Adjust	Indicates whether automatic adjustment for daylight savings time is enabled on the phone.	From Cisco Unified Communications Manager Administration, choose **System > Phone NTP Reference.**

Ethernet Configuration Menu

The Ethernet Configuration menu includes the options that are described in Table 4-16.

Table 4-16 Ethernet Configuration Menu Option

Option	Description	To Change
Span to PC Port (applies to 7911G only)	Indicates whether the phone will forward packets transmitted and received on the network port to the access port. Enable this option if an application that requires monitoring of the phone's traffic is being run on the access port. These applications include monitoring and recording applications (common in call center environments) and network packet capture tools that are used for diagnostic purposes.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Forwarding Delay (applies to 7911G only)	Indicates whether the internal switch begins forwarding packets between the PC port and switched port on the phone when the phone becomes active. • When forwarding delay is set to disabled, the internal switch begins forwarding packets immediately. • When forwarding delay is set to enabled, the internal switch waits 8 seconds before forwarding packets between the PC port and the switch port. Default is disabled.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.

Security Configuration Menu

The Security Configuration menu that you display from the Device Configuration menu displays settings that relate to security for the phone.

Note The phone also has a Security Configuration menu that you access directly from the Settings menu. For information about the security options on that menu, see Security Configuration Menu, page 4-27.

Table 4-17 describes the Security Configuration menu options.

Table 4-17 Security Configuration Menu Options

Option	Description	To Change
PC Port Disabled (applies to 7911G only)	Indicates whether the access port on the phone is enabled (No) or disabled (Yes).	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
GARP Enabled	Indicates whether the phone learns MAC addresses from Gratuitous ARP responses. Disabling the phone's ability to accept Gratuitous ARP will prevent applications that use this mechanism to monitor and record voice streams from working. If voice monitoring is not desired, set this option to No (disabled).	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.

Table 4-17 Security Configuration Menu Options (continued)

Option	Description	To Change
Voice VLAN Enabled (applies to 7911G only)	Indicates whether the phone allows a device attached to the access port to access the Voice VLAN. Setting this option to No (disabled) prevents the attached PC from sending and receiving data on the Voice VLAN. This setting also prevents the PC from receiving data sent and received by the phone. Set this setting to Yes (enabled) if an application that requires monitoring of the phone's traffic is running on the PC. These applications include monitoring and recording applications and network monitoring software.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
Web Access Enabled	Indicates whether web access is enabled (Yes) or disabled (No) for the phone.	For more information, see Disabling and Enabling Web Page Access, page 8-3.
Security Mode	Displays the security mode that is set for the phone.	Use Cisco Unified Communications Manager Administration to modify.
Logging Display	Used by Cisco Technical Assistance Center (TAC) for troubleshooting. The Cisco Unified IP Phone 7911G can be configured for Enabled/Disabled/PC Controlled. The Cisco Unified IP Phone 7906G supports only Enabled/Disabled (no PC Controlled).	—

QoS Configuration Menu

The QoS Configuration menu displays information that relates to quality of service (QoS) for the phone. Table 4-18 describes the QoS Configuration menu options.

Table 4-18 QoS Configuration Menu Options

Option	Description	To Change
DSCP For Call Control	DSCP IP classification for call control signaling.	From Cisco Unified Communications Manager Administration, choose **System > Enterprise Parameters**.
DSCP For Configuration	DSCP IP classification for any phone configuration transfer.	From Cisco Unified Communications Manager Administration, choose **System > Enterprise Parameters**.
DSCP For Services	DSCP IP classification for phone-based services.	From Cisco Unified Communications Manager Administration, choose **System > Enterprise Parameters**.

Related Topics

- Displaying a Configuration Menu, page 4-2
- Network Configuration Menu, page 4-5

Network Configuration

The Network Configuration menu displays device-specific network configuration settings on the phone. Table 4-19 describes the options in this menu.

Note The phone also has a Network Configuration menu that you access from the main menu. For information about the options on that menu, see Network Configuration Menu, page 4-5.

Table 4-19 Network Configuration Menu Options

Option	Description	To Change
Load Server	Used to optimize installation time for phone firmware upgrades and offload the WAN by storing images locally, negating the need to traverse the WAN link for each phone's upgrade. You can set the Load Server to another TFTP server IP address or name (other than the TFTP Server 1 or TFTP Server 2) from which the phone firmware can be retrieved for phone upgrades. When the Load Server option is set, the phone contacts the designated server for the firmware upgrade. **Note** The Load Server option allows you to specify an alternate TFTP server for phone upgrades only. The phone continues to use TFTP Server 1 or TFTP Server 2 to obtain configuration files. The Load Server option does not provide management of the process and of the files, such as file transfer, compression, or deletion.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
IPv6 Load Server	Used to optimize installation time for phone firmware upgrades and offload the WAN by storing images locally, negating the need to traverse the WAN link for each phone's upgrade. You can set the Load Server to another TFTP server IP address or name (other than the IPv6 TFTP Server 1 or IPv6 TFTP Server 2) from which the phone firmware can be retrieved for phone upgrades. When the Load Server option is set, the phone contacts the designated server for the firmware upgrade. **Note** The Load Server option allows you to specify an alternate TFTP server for phone upgrades only. The phone continues to use IPv6 TFTP Server 1 or IPv6 TFTP Server 2 to obtain configuration files. The Load Server option does not provide management of the process and of the files, such as file transfer, compression, or deletion. This feature is disabled in this release.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
RTP Control Protocol	Indicates whether the phone supports the Real Time Control Protocol. Settings include: • Enabled • Disabled—default If this feature is disabled, several call statistic values display as 0. For additional information, see the following sections: • Call Statistics Screen, page 7-12 • Streaming Statistics, page 8-11	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.

Table 4-19 Network Configuration Menu Options (continued)

Option	Description	To Change
CDP: SW Port	Indicates whether CDP is enabled on the switch port (default is enabled). • Enable CDP on the switch port for VLAN assignment for the phone, power negotiation, QoS management, and 802.1x security. • Enable CDP on the switch port when the phone is connected to a Cisco switch. **Note** When CDP is disabled in Cisco Unified Communications Manager, a warning is presented, indicating that CDP should be disabled on the switch port only if the phone is connected to a non-Cisco switch. The current PC and switch port CDP values are shown on the Settings menu.	Use Cisco Unified Communications Manager Administration, and choose **Device > Phone > Phone Configuration**.
Peer Firmware Sharing	The Peer Firmware Sharing feature provides these advantages in high speed campus LAN settings: • Limits congestion on TFTP transfers to centralized TFTP servers • Eliminates the need to manually control firmware upgrades • Reduces phone downtime during upgrades when large numbers of devices are reset simultaneously Peer Firmware Sharing may also aid in firmware upgrades in branch/remote office deployment scenarios over bandwidth-limited WAN links. When enabled, it allows the phone to discover like phones on the subnet that are requesting the files that make up the firmware image, and to automatically assemble transfer hierarchies on a per-file basis. The individual files making up the firmware image are retrieved from the TFTP server by only the root phone in the hierarchy, and are then rapidly transferred down the transfer hierarchy to the other phones on the subnet using TCP connections. This menu option indicates whether the phone supports Peer Firmware Sharing. Settings include: • Enabled • Disabled—default	Use Cisco Unified Communications Manager Administration, and choose **Device > Phone > Phone Configuration**.
Log Server	Indicates the IP address and port of the remote logging machine to which the phone sends log messages. These log messages help in debugging the peer to peer image distribution feature. **Note** The remote logging setting does not affect the sharing log messages sent to the phone log.	Use Cisco Unified Communications Manager Administration, and choose **Device > Phone > Phone Configuration**.

Table 4-19 Network Configuration Menu Options (continued)

Option	Description	To Change
IPv6 Log Server	Indicates the IP address and port of the remote logging machine to which the phone sends log messages. These log messages help in debugging the peer to peer image distribution feature. **Note** The remote logging setting does not affect the sharing log messages sent to the phone log. This feature is disabled in this release.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
CDP: PC Port (applies to 7911G only)	Indicates whether CDP is enabled on the PC port (default is enabled). Enable CDP on the PC port when Cisco VT Advantage/Unified Video Advantage (CVTA) is connected to the PC port. CVTA does not work without CDP interaction with the phone. **Note** When CDP is disabled in Cisco Unified Communications Manager, a warning is displayed, indicating that disabling CDP on the PC port prevents CVTA from working. **Note** The current PC and switch port CDP values are shown on the Settings menu.	Use Cisco Unified Communications Manager Administration, and choose **Device > Phone**.
LLDP: PC Port	Enables and disables Link Layer Discovery Protocol (LLDP) on the PC port. Use this setting to force the phone to use a specific discovery protocol, which should match the protocol supported by the switch. Settings include: • Enabled—default • Disabled	Use Cisco Unified Communications Manager Administration, and choose **Device > Phone > Phone Configuration**
LLDP-MED: SW Port	Enables and disables Link Layer Discovery Protocol Media Endpoint Discovery (LLDP-MED) on the switch port. Use this setting to force the phone to use a specific discovery protocol, which should match the protocol supported by the switch. Settings include: • Enabled—default • Disabled	Use Cisco Unified Communications Manager Administration, and choose **Device > Phone > Phone Configuration**
LLDP Power Priority	Advertises the phone's power priority to the switch, enabling the switch to appropriately provide power to the phones. Settings include: • Unknown—default • Low • High • Critical	Use Cisco Unified Communications Manager Administration, and choose **Device > Phone > Phone Configuration**
LLDP Asset ID	Identifies the asset ID assigned to the phone for inventory management.	Use Cisco Unified Communications Manager Administration, and choose **Device > Phone > Phone Configuration**

Table 4-19 Network Configuration Menu Options (continued)

Option	Description	To Change
IP Addressing Mode	Displays the IP addressing mode that is available on the phone—IPv4 only, IPv6 only, or IPv4 and IPv6.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > Common Device Configuration**.
IP Preference Mode Control	Indicates the IP address version that the phone uses during signaling with Cisco Unified Communications Manager when both IPv4 and IPv6 are both available on the phone. Displays one of the following options: • IPv4—The dual-stack phone prefers to establish a connection via an IPv4 address during a signaling event. • IPv6—The dual-stack phone prefers to establish a connection via an IPv6 address during a signaling event.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > Common Device Configuration**.
Auto IP Configuration	Displays whether the auto configurations is enabled or disabled on the phone. The Auto IP Configuration setting along with the DHCPv6 setting determine how the IP Phone obtains its IPv6 address and other network settings. For more information on how these two settings affect the network settings on the phone, see Table 4-5. **Note** Use the Allow Auto-Configuration for Phones setting in Cisco Unified Communications Manager Administration.	From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > Common Device Configuration**.
IPv6 Load Server	Used to optimize installation time for phone firmware upgrades and off load the WAN by storing images locally, negating the need to traverse the WAN link for each phone's upgrade. You can set the Load Server to another TFTP server IP address or name (other than the IPv6 TFTP Server 1 or IPv6 TFTP Server 2) from which the phone firmware can be retrieved for phone upgrades. When the Load Server option is set, the phone contacts the designated server for the firmware upgrade. **Note** The Load Server option allows you to specify an alternate TFTP server for phone upgrades only. The phone continues to use IPv6 TFTP Server 1 or IPv6 TFTP Server 2 to obtain configuration files. The Load Server option does not provide management of the process and of the files, such as file transfer, compression, or deletion. **Note** When you configure both an IPv6 Load Server and a Load Server (for IPv4), the IPv6 Load server takes precedence.	Use Cisco Unified Communications Manager Administration to modify.
IPv6 Log Server	Indicates the IP address and port of the remote logging machine to which the phone sends log messages. These log messages help in debugging the peer to peer image distribution feature. **Note** The remote logging setting does not affect the sharing log messages sent to the phone log.	Use Cisco Unified Communications Manager Administration to modify.

Related Topics

- Displaying a Configuration Menu, page 4-2
- Network Configuration Menu, page 4-5

Security Configuration Menu

The Security Configuration menu that you access directly from the Settings menu provides information about various security settings. It also provides access to the Trust List menu. This menu indicates if the CTL or ITL file is installed on the phone.

For instructions about how to access the Device Configuration menu and its sub-menus, see Displaying a Configuration Menu, page 4-2.

Note The phone also has a Security Configuration menu that you access from the Device menu. For information about the security options on that menu, see Security Configuration Menu, page 4-27.

Table 4-20 describes the options in this menu.

Table 4-20 Security Configuration Menu Options

Option	Description	To Change
Web Access Enabled	Indicates whether web access is enabled (Yes) or disabled (No) for the phone.	For more information, see Disabling and Enabling Web Page Access, page 8-3.
Security Mode	Displays the security mode that is set for the phone.	From Cisco Unified Communications Manager Administration, choose **Device > Phone > Phone Configuration**.
MIC	Indicates whether a manufacturing installed certificate (used for the security features) is installed on the phone (Yes) or is not installed on the phone (No).	For information about how to manage the MIC for your phone, refer to the Using the Certificate Authority Proxy Function in *Cisco Unified Communications Manager Security Guide*.
LSC	Indicates whether a locally significant certificate (used for the security features) is installed on the phone (Yes) or is not installed on the phone (No).	For information about how to manage the MIC for your phone, refer to Using the Certificate Authority Proxy Function in *Cisco Unified Communications Manager Security Guide*.
Trust List	The Trust List is a top-level menu that provides submenus for the CTL, ITL, and Signed Configuration files. The CTL File submenu displays the contents of the CTL file. The ITL File submenu displays contents of the ITL file. The CTL and ITL files submenus also display the MD5 hash of the file. The MD5 hash value from the phone can be compared with the MD5 hash value of the file from the TFTP server to verify if the correct file is installed on the phone. The Signed Configuration File submenu displays the SRST certificate that is installed via the authenticated digitally signed configuration file.	For more information, see Trust List Menu, page 4-36.

Table 4-20 Security Configuration Menu Options (continued)

Option	Description	To Change
IPv6 CAPF Server	Displays the IP address and the port of the IPv6 CAPF server that the phone uses.	For more information about this server, refer to the Using the Certificate Authority Proxy Function in *Cisco Unified Communications Manager Security Guide*.
802.1X Authentication	Allows you to enable 802.1X authentication for this phone.	See 802.1X Authentication and Status, page 4-37.
802.1X Authentication Status	Displays real-time status progress of the 802.1X authentication transaction.	Display only—Cannot configure.

CTL File Submenu

The CTL File submenu includes the options that are described in Table 4-21.

If a CTL file is installed on the phone, you can access the CTL File submenu by pressing the **Applications Menu** button and choosing **Security Configuration > Trust List**.

Table 4-21 CTL File Settings

Option	Description	To Change
CTL File	Displays the MD5 hash of the CTL file that is installed in the phone. If security is configured for the phone, the CTL file installs automatically when the phone reboots or resets. A locked padlock icon in this option indicates that the CTL file is locked. An unlocked padlock icon indicates that the CTL file is unlocked.	For more information about the CTL file, refer to Configuring the Cisco CTL Client in *Cisco Unified Communications Manager Security Guide*.
Unified CM/ TFTP Server	Common Name (from the Cisco Unified Communications Manager Certificate) of a Cisco Unified Communications Manager and TFTP server used by the phone. Also displays a certificate icon if a certificate is installed for this server. If the certificate of the TFTP (TFTP Server 1) or the backup TFTP (TFTP Server 2) is not in the CTL or ITL file, one of the files must be unlocked.	For information about changing these options, see Network Configuration Menu, page 4-5.
Application Server	Common Name (from the Cisco Unified Communications Manager Certificate) of the trusted application server used by the phone. Also displays a certificate icon. A phone-trust certificate is used to authenticate application servers with which the phone communicates. One Application Server menu item appears for each phone-trust store whose certificates have been uploaded into Cisco Unified OS Administration and later downloaded into the phone's CTL file.	For more information about phone-trust certificates, refer to the following manuals: • *Cisco Unified Communications Operating System Administration Guide*, Security. • *Cisco Unified Communication Manager Security Guide*, Security Overview.

Unlocking the CTL and ITL files

To unlock the CTL and ITL files from the Security Configuration menu, follow these steps:

Procedure

Step 1 Press **# to unlock options on the overall setting menu of the Cisco Unified IP Phone.

Step 2 Select **Trust List** > **CTL** or **ITL file** (depending on which file is installed in your phone).

Note If both CTL and ITL files are installed in your phone, you can choose any one of the options.

Step 3 Press **Unlock** softkey to unlock Trust List files on the phone. The CTL or ITL files, if installed on your phone, will be unlocked together.

Note When you press the **Unlock** softkey, it changes to **Lock**. If you decide not to change the TFTP server option, press the **Lock** softkey to lock the CTL file.

ITL File Submenu

The ITL File screen includes the options that are described in Table 4-22.

If an ITL file is installed on the phone, you can access the ITL File submenu by pressing the Settings button and choosing **Security Configuration** > **Trust List**.

Table 4-22 ITL File Settings

Option	Description	To Change
ITL File	Displays the MD5 hash of the ITL file that is installed in the phone. If security is configured for the phone, the ITL file installs automatically when the phone reboots or resets. A locked padlock icon in this option indicates that the ITL file is locked. An unlocked padlock icon indicates that the ITL file is unlocked.	For more information about the CTL file, refer to the Configuring the Cisco ITL Client section in *Cisco Unified Communications Manager Security Guide*.
CAPF Server	Common Name (from the Cisco Unified Communications Manager Certificate) of the CAPF used by the phone. Also displays a certificate icon if a certificate is installed for this server.	For more information about this server, refer to the Using the Certificate Authority Proxy Function in *Cisco Unified Communications Manager Security Guide*.
Unified CM/TFTP Server	Common Name (from the Cisco Unified Communications Manager Certificate) of a Cisco Unified Communications Manager and TFTP server used by the phone. Also displays a certificate icon if a certificate is installed for this server. If neither the certificate of TFTP (TFTP Server 1) nor the certificate of backup TFTP (TFTP Server 2) is not in the CTL or ITL file, you must unlock the CTL file.	For information about changing these options, see Network Configuration Menu, page 4-5.

Table 4-22 ITL File Settings (continued)

Option	Description	To Change
Application Server	Common Name (from the Cisco Unified Communications Manager Certificate) of the trusted application server used by the phone. Also displays a certificate icon. A phone-trust certificate is used to authenticate application servers with which the phone communicates. One Application Server menu item appears for each phone-trust store whose certificates have been uploaded into Cisco Unified OS Administration and later downloaded into the phone's ITL file.	For more information about phone-trust certificates, refer to the following manuals: *Cisco Unified Communications Operating System Administration Guide*, Security. *Cisco Unified Communication Manager Security Guide*, Security Overview.
Trust Verification Service (TVS) Server	Common Name (from the Cisco Unified Communications Manager Certificate) of the trusted application server used by the phone. Also displays a certificate icon. A phone-trust TVS certificate is used to authenticate TVS servers with which the phone communicates. There can be more than one entry for the TVS servers.	For more information, refer to the *Cisco Unified Communications Manager System Administrator Guide.*

Trust List Menu

The Trust List menu provides a top-level menu containing CTL, ITL, and the Signed Configuration submenus. The content of the Signed Configuration file is SRST.

The Trust List menu displays information about all of the servers that the phone trusts. Table 4-23 describes the options in this menu.

Table 4-23 Trust List Information

Option	Description	To Change
CAPF Server	Common Name (from the Cisco Unified Communications Manager Certificate) of the CAPF server used by the phone. Also displays a certificate icon if a certificate is installed for this server.	For more information about these settings, refer to Configuring the Cisco CTL Client in *Cisco Unified Communications Manager Security Guide*.
Unified CM/ TFTP Server	Common Name (from the Cisco Unified Communications Manager Certificate) of a Cisco Unified Communications Manager and the TFTP server used by the phone.Also displays a certificate icon if a certificate is installed for this server.	For more information about these settings, refer to Configuring the Cisco CTL Client in *Cisco Unified Communications Manager Security Guide*.

Table 4-23 Trust List Information (continued)

Option	Description	To Change
SRST Router	Common Name (from the Cisco Unified Communications Manager Certificate) of the trusted SRST router that is available to the phone, if such a device has been configured in Cisco Unified Communications Manager Administration. Also displays a certificate icon if a certificate is installed for this server.	For more information about these settings, refer to Configuring the Cisco CTL Client in *Cisco Unified Communications Manager Security Guide*.
Application Server	Common Name (from the Cisco Unified Communications Manager Certificate) of the trusted application server used by the phone. Also displays a certificate icon. A phone-trust certificate is used to authenticate application servers with which the phone communicates. One Application Server menu item appears for each phone-trust store whose certificates have been uploaded into Cisco Unified OS Administration and later downloaded into the Cisco Unified IP Phone CTL file.	For more information about phone-trust certificates, refer to the following manuals: • *Cisco Unified Communications Operating System Administration Guide*, Security. • *Cisco Unified Communication Manager Security Guide*, Security Overview.

802.1X Authentication and Status

Use the options that are described in the following tables to enable 802.1X authentication and monitor its progress:

- Table 4-24 on page 4-38—802.1X Authentication Settings
- Table 4-25 on page 4-39—802.1X Authentication Real-Time Status

Table 4-24 802.1X Authentication Settings

Option	Description	To Change
Device Authentication	Determines whether 802.1X authentication is enabled: • **Enabled**—Phone uses 802.1X authentication to request network access. • **Disabled**—Default setting in which the phone uses CDP to acquire VLAN and network access.	1. Choose **Settings > Security Configuration > 802.1X Authentication > Device Authentication**. 2. Set the Device Authentication option to Enabled or Disabled. 3. Press the **Save** softkey.
EAP-MD5	Specifies a password for use with 802.1X Authentication using the following menu options (described in the following rows): • Device ID • Shared Secret • Realm	Choose **Settings > Security Configuration > 802.1X Authentication > EAP-MD5**.
	Device ID—A derivative of the phone model number and unique MAC Address displayed in this format: CP-<model>-SEP-<MAC Address>	Display only—Cannot configure.
	Shared Secret—Choose a password to use on the phone and on the authentication server. The password must be between 6 and 32 characters, consisting of any combination of numbers or letters. **Note** If you disable 802.1X authentication or perform a factory reset of the phone, the shared secret is deleted.	1. Choose **EAP-MD5 > Shared Secret**. 2. Enter the shared secret. 3. Press **Save**. See Troubleshooting Cisco Unified IP Phone Security, page 9-8 for assistance in recovering from a deleted shared secret.
	Realm—Indicates the user network domain, always set as *Network*.	Display only—Cannot configure.

Table 4-25 802.1X Authentication Real-Time Status

Option	Description	To Change
802.1X Authentication Status	Real-time progress of the 802.1X authentication status. Displays one of the following states: • Disabled—802.1X is disabled and transaction was not attempted • Disconnected—Physical link is down or disconnected • Connecting—Trying to discover or acquire the authenticator • Acquired—Authenticator acquired, awaiting authentication to begin • Authenticating—Authentication in progress • Authenticated—Authentication successful or implicit authentication due to timeouts • Held—Authentication failed, waiting before next attempt (approximately 60 seconds)	Display only—Cannot configure.

CHAPTER 5

Configuring Features, Templates, Services, and Users

After you install Cisco Unified IP Phones in your network, configure their network settings, and add them to Cisco Unified Communications Manager, you must use Cisco Unified Communications Manager Administration to configure communications features, optionally modify phone templates, set up services, and assign users.

This chapter provides an overview of these configuration and setup procedures. Cisco Unified Communications Manager documentation provides detailed instructions for these procedures.

For suggestions about how to provide users with information about features, and what information to provide, see Providing Information to Users, page A-1

For information about setting up phones in non-English environments, see Appendix C, Supporting International Users

This chapter includes following topics:

- Telephony Features Available for the Cisco Unified IP Phone, page 5-1
- Configuring Product Specific Configuration Parameters, page 5-15
- Configuring Corporate and Personal Directories, page 5-16
- Modifying Phone Button Templates, page 5-18
- Configuring Softkey Templates, page 5-19
- Setting Up Services, page 5-20
- Adding Users to Cisco Unified Communications Manager, page 5-21
- Managing the User Options Web Pages, page 5-21

Telephony Features Available for the Cisco Unified IP Phone

After you add Cisco Unified IP Phones to Cisco Unified Communications Manager, you can add functionality to the phones. Table 5-1 includes a list of supported telephony features, many of which you configure by using Cisco Unified Communications Manager Administration. The Configuration Reference column lists Cisco Unified Communications Manager documentation that contains configuration procedures and related information.

For more information about using most of these features on the phone, refer to the *Cisco Unified IP Phones 7906G and 7911G Guide*.

Note Cisco Unified Communications Manager Administration also provides several service parameters that you can use to configure various telephony functions. For more information on accessing and configuring service parameters, refer to *Cisco Unified Communications Manager Administration Guide*. For more information on the functions of a service, click on the name of the parameter or the question mark help button in the Service Parameter Configuration window.

Table 5-1 Telephony Features for the Cisco Unified IP Phone

Feature	Description	Configuration Reference
Abbreviated dialing	A user can configure up to 99 speed-dial entries. Speed-dial entries that are not assigned to the speed-dial buttons on the phone are used for abbreviated dialing. When a user starts dialing digits, the **AbbrDial** softkey appears, and the user can access any speed-dial entry by entering the appropriate index. **Note** You can use Abbreviated Dialing while on-hook or off-hook.	For more information, refer to: • *Cisco Unified Communications Manager Administration Guide,* Cisco Unified IP Phone Configuration. • *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phones.
Add Select to Join	Creates a conference by joining together existing calls that are on a single phone line.	For more information, refer to the *Cisco Unified IP Phone Guide*, Basic Call Handling chapter, Making Conference Calls section.
Any Call Pickup	Allows users to pick up a redirected call via the CTI application, on any line in their call pickup group, regardless of how the call was routed to the phone.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Call Pickup.
Audible Message Waiting Indicator	A stutter tone from the handset, headset, or speakerphone indicates that a user has one or more new voice messages on a line. **Note** The stutter tone is line-specific. You hear it only when using the line with the waiting messages.	For more information, refer to: • *Cisco Unified Communications Manager Administration Guide,* Message Waiting Configuration. • *Cisco Unified Communications Manager System Guide,* Voice Mail Connectivity to Cisco Unified Communications Manager.
Auto answer	Causes the speakerphone to go off hook automatically when an incoming call is received. The user can monitor the call using the speaker but must pick up the handset to speak to the caller.	For more information, refer to the *Cisco Unified Communications Manager Administration Guide,* Configuring Directory Numbers.
Auto dial	Allows the phone user to choose from matching numbers in the Placed Calls log while dialing. To place the call, the user can choose a number from the Auto Dial list or continue to enter digits manually.	

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Barge (and cBarge)	Allows a user to join an in-progress call on a shared line. Phones support Barge in two conference modes: • Built-in conference bridge at the target device (the phone that is being barged). This mode uses the **Barge** softkey. • Shared conference bridge. This mode uses the **cBarge** softkey.	For more information, refer to: • *Cisco Unified Communications Manager Administration Guide,* Cisco Unified IP Phone Configuration. • *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phones. • *Cisco Unified Communications Manager Features and Services Guide,* Barge and Privacy.
Block external to external transfer	Prevents users from transferring an external call to another external number.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, External Call Transfer Restrictions.
Call Chaperone	Allows an authorized Chaperone user to supervise and record a call. The Call Chaperone user intercepts and answers the call from calling party, manually creates a conference to the called party, and remains on the conference to supervise and record the call. Cisco Unified IP Phones that have the Call Chaperone feature configured on them have a Record softkey. The Call Chaperone user presses the Record softkey to record a call. For chaperoned calls, an announcement is played or spoken by one of the participants at the start of the call. An announcement will alert later participants in the call that the call is being recorded. The Call Chaperone feature is supported only with External Call Control, which allows Cisco Unified Communications Manager to route audio and video calls to a route server that hosts routing rules.	
Call display restrictions	Determines the information that will display for calling or connected lines, depending on the parties who are involved in the call.	For more information, refer to: • *Cisco Unified Communications Manager Administration Guide*, Cisco Unified IP Phone Configuration. • *Cisco Unified Communications Manager System Guide,* Understanding Route Plans. • *Cisco Unified Communications Manager Features and Services Guide,* Call Display Restrictions.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Call forward	Allows users to redirect incoming calls to another number. Call forward options include Call Forward All, Call Forward Busy, Call Forward No Answer, and Call Forward No Coverage.	For more information, refer to: • *Cisco Unified Communications Manager Administration Guide,* Directory Number Configuration. • *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phones.
Call forward all loop breakout	Detects and prevents Call Forward All loops. When a loop is detected, the Call Forward All configuration is ignored and the call rings through.	For more information, refer to the *Cisco Unified Communications Manager System Guide*, Cisco Unified IP Phones.
Call forward all loop prevention	Prevents a user from configuring a Call Forward All destination directly on the phone that creates a Call Forward All loop or that creates a forward chain with more hops than the existing Forward Maximum Hop Count service parameter allows.	For more information, refer to the *Cisco Unified Communications Manager System Guide*, Cisco Unified IP Phones.
Call forward destination override	Allows you to override Call Forward All (CFA) in cases where the CFA target places a call to the CFA initiator. This feature allows the CFA target to reach the CFA initiator for important calls. The override works whether the CFA target phone number is internal or external.	For more information, refer to *Cisco Unified Communications Manager System Guide*, Understanding Directory Numbers.
Call park	Places the call on hold so that anyone connected to the Cisco Unified Communications Manager system can retrieve the call. **Note** If you are using the Park softkey, avoid configuring the Directed Call Park feature. This prevents users from confusing the two Call Park features.	For more information, refer to: • *Cisco Unified Communications Manager Administration Guide,* Call Park. • *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phones. • *Cisco Unified Communications Manager Features and Services Guide*, Call Park and Directed Call Park.
Call pickup	Allows users to redirect a call that is ringing on another phone within their pickup group to their phone. You can configure an audio and/or visual alert for the primary line on the phone. This alert notifies the users that a call is ringing in their pickup group.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Call Pickup Configuration.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Call recording	Enables recording of an active call. The user might hear a recording audible alert tone during a call when it is being recorded. When a call is secured, the security status of the call is displayed as a lock icon on Cisco Unified IP Phones. The connected parties might also hear an audible alert tone that indicates the call is secured and is being recorded. **Note** When an active call is being monitored or recorded, you can receive or place intercom calls; however, if you place an intercom call, the active call will be put on hold, which causes the recording session to terminate and the monitoring session to suspend. To resume the monitoring session, the party whose call is being monitored must resume the call.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide,* Monitoring and Recording.
Call waiting	Receives a second incoming call on the same line without disconnecting the first call.	For more information, refer to the *Cisco Unified Communications System Guide*, Understanding Directory Numbers.
Caller ID	Displays the telephone number and name of the caller.	For more information, refer to: • *Cisco Unified Communications Manager Administration Guide,* Configuring Directory Number Configuration. • *Cisco Unified Communications Manager System Guide,* Understanding Route Plans. • *Cisco Unified Communications Manager Features and Services Guide*, Call Display Restrictions.
Caller ID Blocking	Blocks a users phone number or e-mail address.	For more information, refer to: • *Cisco Unified Communications Manager System Guide*, Understanding Route Plans. • *Cisco Unified Communications Manager Administration Guide*, Directory Number Configuration. • Cisco Unified *Communications Manager Administration Guide*, SIP Profile Configuration.
Calling Party Normalization	Globalizes or localizes the incoming calling party number so that the appropriate calling number presentation displays on the phone. Supports the international escape character +.	For more information, refer to the *Cisco Unified Communications Features and Services Guide*, Calling Party Normalization.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Cisco Call Back	Allows a user to receive call back notification on a Cisco Unified IP Phone when a called party becomes available.	For more information, refer to: • *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phone Configuration. • *Cisco Unified Communications Manager Features and Services Guide*, Cisco Call Back.
Cisco Extension Mobility	Enables users to sign into their directory number from any Cisco Unified IP Phone.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide,* Cisco Extension Mobility.
Cisco Extension Mobility Change PIN	Enables a user to change the PIN from a Cisco Unified IP Phone. The PIN can be changed by: • Using the Change Credentials service of a Cisco Unified IP Phone • Using the **ChangePIN** softkey on the Extension Mobility logout screen	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Cisco Extension Mobility.
Cisco Extension Mobility Cross Cluster	Enables a user configured in one cluster to log into a Cisco Unified IP Phone in another cluster. Users from a home cluster log into a Cisco Unified IP Phone at a visiting cluster. **Note** Even though the Intercom feature works with Cisco Extension Mobility (EM), it cannot be used with EMCC because the feature must be enabled with a real phone device. The Intercom feature cannot be enabled with EM profiles.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Cisco Extension Mobility Cross Cluster.
Client matter codes (CMC) (SCCP phones only)	Enables a user to specify that a call relates to a specific client matter. **Note** If you are using this feature, you must disable Enbloc dialing. See Enbloc Dialing for details.	For more information, refer to: • *Cisco Unified Communications Manager Administration Guide,* Client Matter Codes. • *Cisco Unified Communications Manager Features and Services Guide,* Client Matter Codes and Forced Authorization Codes.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Conference	• Allows a user to talk simultaneously with multiple parties by calling each participant individually. Conference features include Conference, Join, cBarge, and Meet-Me. • Allows a non-initiator in a standard (ad hoc) conference to add or remove participants; also allows any conference participant to join together two standard conferences on the same line.	For more information, refer to • *Cisco Unified Communications Manager System Guide*, Cisco Unified IP Phones and Conference Bridges. • The service parameter, Advance Adhoc Conference, (disabled by default in Cisco Unified Communications Manager Administration) allows you to enable these features. **Note** Be sure to inform your users whether these features are activated.
Computer Telephony Integration (CTI) Applications	A computer telephony integration (CTI) route point can designate a virtual device to receive multiple, simultaneous calls for application-controlled redirection.	For more information, refer to the *Cisco Unified Communications Manager Administration Guide,* CTI Route Point Configuration.
Directed Call Park	Allows a user to direct an active call to an available directed call park number. After pressing Transfer, the user dials the directed call park number to store the call.	For more information refer to *Cisco Unified Communications Manager Features and Services Guide,* Call Park and Directed Call Park.
Directed call Pickup	Allows a user to answer a call that is ringing on a particular directory number.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Call Pickup.
Direct transfer	Joins two established calls (calls that are on hold or in connected state) into one call and drops the feature initiator from the call. Does not initiate a consultation call and does not put the active call on hold.	For more information, refer to the *Cisco Unified Communications Manager System Guide*, Cisco Unified IP Phones.
Enbloc Dialing (SCCP phones only)	Enbloc dialing enables SCCP to send all digits of a phone number simultaneously. This feature must be disabled if either Forced Authorization Codes (FAC) or Client Matter Codes (CMC) dialing is being used.	To disable enbloc dialing, in Cisco Unified Communications Manager Administration, go to **Device > Phone.** On the Phone Configuration window, in the "Product Specific Configuration Layout" area, uncheck the "Enbloc Dialing" check box, then click **Apply Config**, then click **Save**.
Distinctive Ring	Users can customize how their phone indicates an incoming call and a new voice mail message.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Custom Phone Rings.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Do Not Disturb (DND)	When DND is turned on, either no audible rings occur during the ringing-in state of a call, or no audible or visual notifications of any type occur. You can configure the phone to have a softkey template with a DND softkey. The following DND-related parameters are configurable in Cisco Unified Communications Manager Administration: • Do Not Disturb—This checkbox allows you to enable DND on a per-phone basis. Choose **Device > Phone > Phone Configuration**. • DND Option—Choose "Call Reject" (to turn off all audible and visual notifications), or "Ringer Off" (to turn off only the ringer). *DND Option* appears on both the Common Phone Profile window and the Phone Configuration window (Phone Configuration window value takes precedence). • DND Incoming Call Alert—Choose the type of alert, if any, to play on a phone for incoming calls when DND is active. This parameter is located on both the Common Phone Profile window and the Phone Configuration window. (Phone Configuration window value takes precedence).	*Cisco Unified Communications Manager Features and Services Guide*, Do Not Disturb.
Fast Dial Service	Allows a user to enter a Fast Dial code to place a call. Fast Dial codes can be assigned to phone numbers or Personal Address Book entries. (See Services in this table.)	For more information, refer to the Modifying a Phone Button Template for Personal Address Book or Fast Dials, page 5-18
Forced authorization codes (FAC) (SCCP phones only)	Controls the types of calls that certain users can place. **Note** If you are using this feature, you must disable Enbloc dialing. See Enbloc Dialing for details.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide,* Client Matter Codes and Forced Authorization Codes.
Group call pickup	Allows a user to answer a call that is ringing on a directory number in another group.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Call Pickup.
Help system	Provides a comprehensive set of topics that appear on the phone screen.	Requires no configuration.
Hold/Resume	Allows the user to move a connected call between an active state and a held state.	• Requires no configuration, unless you want to use music on hold. See Music on hold in this table for information. • See also: Hold Reversion in this table.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Hold Reversion	Limits the amount of time that a call can be on hold before reverting back to the phone that put the call on hold and alerting the user. Reverting calls are distinguished from incoming calls by a single ring (or beep, depending on the new call indicator setting for the line). This notification repeats at intervals if not resumed. A call that triggers Hold Reversion also displays an animated icon in the call bubble and a brief message on the status line. You can configure call focus priority to favor incoming or reverting calls.	For more information about configuring this feature, refer to *Cisco Unified Communications Manager Features and Services Guide,* Hold Reversion.
Hold Status	Enables phones with a shared line to distinguish between the local and remote lines that placed a call on hold.	No configuration is required.
Hunt Group Display	Provides load sharing for calls to a main directory number. A hunt group contains a series of directory numbers that can answer the incoming calls. When an incoming call is offered to a directory number that is part of the hunt group, this feature displays the main directory number in addition to the calling party.	For more information, refer to: • *Cisco Unified Communications Manager Administration Guide,* Hunt Group Configuration. • *Cisco Unified Communications Manager System Guide,* Understanding Route Plans. • *Cisco Unified Communications Manager Administration Guide*, CTI Route Point Configuration.
Immediate Divert	Allows a user to transfer a ringing, connected, or held call directly to a voice-messaging system.When a call is diverted, the line becomes available to make or receive new calls.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Immediate Divert.
Immediate Divert—Enhanced	Allows users to transfer incoming calls directly to their voice messaging system or to the voice messaging system of the original called party.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Immediate Divert.
Intelligent Session Control	Reroutes a direct call to users' mobile phone to the enterprise number (desk phone). For an incoming call to remote destination (mobile phone), only remote destination rings; desk phone does not ring. When the call is answered on their mobile phone, the desk phone displays a Remote In Use message. During these calls, users can make use of various features of their mobile phone.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Cisco Unified Mobility.
Join	Allows users to initiate an ad hoc conference by using the **Join** softkey. Join does not create a consultation call and does not put the active call on hold. Join can include more than two calls, which results in a call with more than three parties.	For more information, refer to the *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phones.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Log out of hunt groups	Allows users to log out of a hunt group and temporarily block calls from ringing their phone when they are not available to take calls. Logging out of hunt groups does not prevent non-hunt group calls from ringing their phone.	For more information refer to the *Cisco Unified Communications Manager System Guide,* Understanding Route Plans.
Malicious call identification (MCID)	Allows you to report a call of a malicious nature by requesting that Cisco Unified Communications Manager identify and register the source of an incoming call in the network.	For more information refer to: • *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phones. • *Cisco Unified Communications Manager Features and Services Guide,* Malicious Call Identification.
Meet-Me conference	Enables other callers to join in a conference.	For more information refer to: • *Cisco Unified Communications Manager Administration Guide*, Meet-Me Number/Pattern Configuration. • *Cisco Unified Communications Manager System Guide*, Conference Bridges.
Message waiting	Indicates that one or more voice messages are waiting for a user.	For more information refer to: • *Cisco Unified Communications Manager Administration Guide,* Message Waiting Configuration. • *Cisco Unified Communications Manager System Guide*, Voice Mail Connectivity to Cisco Unified Communications Manager.
Missed call logging	Allows a user to specify whether missed calls will be logged in the missed calls directory for a given line appearance.	For more information refer to the *Cisco Unified Communications Manager Administration Guide*, Directory Number Configuration.
Mobile Connect	Enables users to manage business calls using a single phone number and pick up in-progress calls on the desktop phone and mobile phone. Users can restrict the group of callers according to phone number and time of day.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Cisco Unified Mobility.
Mobile Voice Access	Extends Mobile Connect capabilities by allowing users to access an interactive voice response (IVR) system to originate a call from a remote device such as a cellular phone.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Cisco Unified Mobility.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Multilevel Precedence and Preemption (MLPP) (SCCP phones only)	Allows properly validated users to place priority calls. If necessary, users can preempt lower-priority phone calls. Also allows the use of the call-forward alternate party (CFAP) feature for forwarding a precedence call.	For more information refer to the *Cisco Unified Communications Manager Features and Services Guide,* Multilevel Precedence and Preemption.
Multiple Calls Per Line Appearance	Each line can support multiple calls. Only one call can be active at any time; other calls are automatically placed on hold.	For more information refer to the *Cisco Unified Communications Manager Administration Guide*, Cisco Unified IP Phone Configuration.
Music on hold	Plays music while callers are on hold.	For more information refer to the *Cisco Unified Communications Manager Features and Services Guide*, Music On Hold.
Mute	Mutes the microphone from the handset or headset.	Requires no configuration.
Onhook call transfer	Allows a user to press a single Transfer softkey and then go onhook to complete a call transfer.	For more information refer to the *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phones.
Onhook predialing	Allows a user to dial a number without going off hook. The user can then either pick up the handset or press the Dial softkey.	For more information, refer to the *Cisco Unified IP Phone 7962G Phone Guide*, Basic Call Handling chapter.
Other group pickup	Allows a user to answer a call that is ringing on a phone in another group that is associated with the user's group.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Call Pickup.
Phone secure web access	Cisco Unified IP Phones can now securely access the web with the use of a phone trust store called “phone-trust.”	For more information refer to the *Cisco Unified Communication Manager Security Guide,* Security Overview.
Private Line Automated Ringdown (PLAR)	The Cisco Unified Communications Manager administrator can configure a phone number that the Cisco Unified IP Phone dials as soon as the handset goes off hook. This can be useful for phones that are designated for calling emergency or “hotline” numbers.	For more information refer to the Configuring PLAR section in the Directory Number Configuration in the *Cisco Unified Communications Manager Administration Guide*.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Privacy	Prevents users who share a line from adding themselves to a call and from viewing information on their phone screens about the call of the other user.	For more information refer to: • *Cisco Unified Communications Manager Administration Guide*, Cisco Unified IP Phone Configuration. • *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phones. • *Cisco Unified Communications Manager Features and Services Guide* Barge and Privacy.
Programmable line keys (PLK)	The administrator can assign features to line buttons. Softkeys normally control these features; for example, New Call, Call Back, End Call, and Forward All. When the administrator configures these features on the line buttons, they always remain visible, so users can have a “hard” New Call key.	For more information, refer to: • *Cisco Unified Communications Manager System Guide*, Cisco Unified IP Phones. • *Cisco Unified Communications Manager Administration Guide*, Phone Button Template Configuration. • *Cisco Unified Communications Manager Administration Guide*, Modifying Phone Button Templates
Protected calling	Provides a secure (encrypted) connection between two phones. A security tone is played at the beginning of the call to indicate that both phones are protected. Some features, such as conference calling, shared lines, Extension Mobility, and Join Across Lines are not available when protected calling is configured. Protected calls are not authenticated.	For more information about security, see Overview of Supported Security Features, page 1-11. For additional information, refer to *Cisco Unified Communications Manager Security Guide.*
Quality Reporting Tool (QRT)	Allows users to use the **QRT** softkey on a phone to submit information about problem phone calls. QRT can be configured for either of two user modes, depending upon the amount of user interaction desired with QRT.	For more information refer to: • *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phones. • *Cisco Unified Communications Manager Features and Services Guide,* Quality Report Tool.
Redial	Redials the last number dialed on the Cisco Unified IP Phone.	Requires no configuration.
Ring setting	Identifies ring type used for a line when a phone has another active call	For more information refer to the *Cisco Unified Communications Manager Administration Guide*, Directory Number Configuration.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Ringer Volume Control	The Ringer Volume Control feature enables the system administrator to control the minimum ringer-volume setting and adjust the minimum volume level for the ringer. Individual users cannot make the changes to the minimum ringer-volume setting. When a user presses the minus (–) side of the Volume button to reduce the ringer volume in an on-hook state, the volume decreases only to the configured minimum volume-level setting. When the minimum volume level is reached, no status message appears. After a system restart, the minimum ringer volume resets to the minimum ringer-volume setting that is received from the configuration file. If the system administrator configured a new minimum volume level since the last startup and the end user had previously set the minimum ringer volume lower, the ringer volume will be set to the minimum value from the configuration file, not to the user's setting. This feature does not apply to handset, speaker, and headset volumes during calls.	To configure the parameter in the Cisco Unified CM Administration application, choose **Device > Phone**, select the appropriate IP phones, and scroll to the Product Specific Configuration Layout pane.
Secure Conference	• Allows secure phones to place conference calls by using a secured conference bridge. • As new participants are added by using Confrn, Join, cBarge, Barge softkeys or MeetMe conferencing, the secure call icon displays as long as all participants use secure phones. • The Conference List displays the security level of each conference participant. Initiators can remove nonsecure participants from the Conference List. (Non-initiators can add or remove conference participants if the AdvanceAdhocConference parameter is set.)	For more information about security, see Overview of Supported Security Features, page 1-11. For additional information, refer to these: • *Cisco Unified Communications Manager System Guide*, Conference Bridges. • *Cisco Unified Communications Manager Administration Guide,* Conference Bridge Configuration. • *Cisco Unified Communications Manager Security Guide*.
Services	Allows you to use the Cisco Unified IP Phone Services Configuration menu in Cisco Unified Communications Manager Administration to define and maintain the list of phone services to which users can subscribe.	For more information refer to: • *Cisco Unified Communications Manager Administration Guide,* Cisco Unified IP Phone Configuration. • *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phone Services.
Services URL button	Provides one-touch access to information services.	For more information refer to the *Cisco Unified Communications Manager Administration Guide* for configuration procedures.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Session Handoff	Allows users to switch calls from a mobile phone to Cisco Unified devices that share the same line. Handsets on all the devices on the shared line flash simultaneously. After a user answers the call from one of the Cisco Unified devices, the other Cisco Unified devices that share the same line display a Remote in Use message. However, if the call fails to switch from the mobile phone, the mobile phone may display a Cannot Move Conversation message.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide*, Cisco Unified Mobility and Cisco Unified Mobility Advantage and Cisco Unified Mobile Communicator Integration.
Shared line	Allows a user to have multiple phones that share the same phone number or allows a user to share a phone number with a coworker.	For more information, refer to the *Cisco Unified Communications Manager System Guide*, Understanding Directory Numbers.
Silent Monitoring	Allows a supervisor to silently monitor an active call. The supervisor cannot be heard by either party on the call. The user might hear a monitoring audible alert tone during a call when it is being monitored. When a call is secured, the security status of the call is displayed as a lock icon on Cisco Unified IP Phones. The connected parties might also hear an audible alert tone that indicates the call is secured and is being monitored. **Note** When an active call is being monitored or recorded, you can receive or place intercom calls; however, if you place an intercom call, the active call will be put on hold, which causes the recording session to terminate and the monitoring session to suspend. To resume the monitoring session, the party whose call is being monitored must resume the call.	For more information, refer to the *Cisco Unified Communications Manager Features and Services Guide,* Monitoring and Recording.
Speed-dial	Dials a specified number that has been previously stored.	For more information refer to: • *Cisco Unified Communications Manager Administration Guide,* Cisco Unified IP Phones Configuration. • *Cisco Unified Communications Manager System Guide,* Cisco Unified IP Phones Configuration.
Time-of-Day Routing	Restricts access to specified telephony features by time period.	For more information refer to: • *Cisco Unified Communications Manager Administration Guide,* Time Period Configuration. • *Cisco Unified Communications Manager System Guide,* Time-of-Day Routing.

Table 5-1 Telephony Features for the Cisco Unified IP Phone (continued)

Feature	Description	Configuration Reference
Time Zone Update	Updates the Cisco Unified IP Phone with time zone changes.	For more information, refer to the *Cisco Unified Communications Manager Administration Guide*, Date/Time Group Configuration.
Voice messaging system	Enables callers to leave voice messages if calls are unanswered.	For more information refer to: • *Cisco Unified Communications Manager Administration Guide,* Cisco Voice-Mail Port Configuration. • *Cisco Unified Communications Manager System Guide*, Voice Mail Connectivity to Cisco Unified Communications Manager.
Video mode (7911G only)	Allows a user to select the video display mode for viewing a video conference, depending on the modes configured in the system.	For more information refer to: • *Cisco Unified Communications Manager Administration Guide*, Conference Bridge Configuration. • *Cisco Unified Communications Manager System Guide*, Understanding Video Telephony.
Video support (7911G only)	Enable video support on the phone.	For more information refer to: • *Cisco Unified Communications Manager Administration Guide*, Conference Bridge Configuration. • *Cisco Unified Communications Manager System Guide,* Understanding Video Telephony. • *Cisco VT Advantage Administration Guide*, Overview of Cisco VT Advantage.

Configuring Product Specific Configuration Parameters

Cisco Unified Communications Manager Administration allows you to set some product specific configuration parameters for Cisco Unified IP Phones. Table 5-2 lists the configuration windows, their path, and the parameters in Cisco Unified Communications Manager Administration.

Table 5-2 Configuration Parameters for Cisco Unified IP Phone

Configuration Window	Path	Parameters
Enterprise Phone Configuration window	**System > Enterprise Phone Configuration**	You can set the following parameters in any of the three configuration windows: • Settings Access • Video Capabilities • Web Access • Load Server • RTCP • Peer Firmware Sharing • Cisco Discovery Protocol (CDP): Switch Port • Cisco Discovery Protocol (CDP): PC Port • Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port • Link Layer Discovery Protocol (LLDP): PC Port • IPv6 Load Server • 802.1x Authentication **Note** When you set the parameters, select the Override Common Settings checkbox for each setting you wish to update. If you do not check this box, the corresponding parameter setting does not take effect. If you set the parameters at the three configuration windows, the setting takes precedence in the following order: • Phone Configuration window • Common Phone Profile window • Enterprise Phone Configuration window

Configuring Corporate and Personal Directories

The **Directories** menu on the Cisco Unified IP Phone 7906G and 7911G gives users access to several directories. These directories can include:

- Corporate Directory—Allows a user to look up phone numbers for co-workers.

  To support this feature, you must configure corporate directories. See Configuring Corporate Directories, page 5-17 for more information.

- Personal Directory—Allows a user to store a set of personal numbers.

  To support this feature, you must provide the user with software to configure the personal directory. See Configuring Personal Directory, page 5-17 for more information.

After the LDAP directory configuration completes, users can use the Corporate Directory service on your Cisco Unified IP Phone to look up users in the corporate directory.

Configuring Corporate Directories

Cisco Unified Communications Manager uses a Lightweight Directory Access Protocol (LDAP) directory to store authentication and authorization information about users of Cisco Unified Communications Manager applications that interface with Cisco Unified Communications Manager. Authentication establishes a user's right to access the system. Authorization identifies the telephony resources that a user is permitted to use, such as a specific telephone extension.

For more information on directories, refer to the *Cisco Unified Communications Manager System Guide*, Understanding Directory chapter.

After completing the LDAP directory configuration, users can use the Corporate Directory service on their Cisco Unified IP Phone to look up users in the corporate directory.

Configuring Personal Directory

Personal Directory consists of the following features:

- Personal Address Book (PAB)
- Personal Fast Dials (Fast Dials)
- Address Book Synchronization Tool (TABSync)

Users can access Personal Directory features by these methods:

- From a web browser—Users can access the PAB and Fast Dials features from the Cisco Unified CM User Options
- From the Cisco Unified IP Phone—Users can choose **Directories > Personal Directory** to access the PAB and Fast Dials features from their phones
- From a Microsoft Windows application—Users can use the TABSync tool to synchronize their PABs with Microsoft Windows Address Book (WAB). Customers who want to use the Microsoft Outlook Address Book (OAB) should begin by importing the data from the OAB into the Windows Address Book (WAB). TabSync can then be used to synchronize the WAB with Personal Directory.

To ensure that Cisco Unified IP Phone Address Book Synchronizer users have access only to end-user data that pertains to them, activate the Cisco UXL Web Service in Cisco Unified Serviceability.

To configure Personal Directory from a web browser, users must access their User Options web pages. You must provide users with a URL and login information.

To synchronize with Microsoft Outlook, users must install the TABSync utility, which is provided by you. To obtain the TABSync software to distribute to users, choose **Application > Plugins** from Cisco Unified Communications Manager Administration, then locate and click **Cisco Unified IP Phone Address Book Synchronizer**.

Modifying Phone Button Templates

Phone button templates let you assign features to phone buttons. On the Cisco Unified IP Phones 7906G and 7911G, only the Privacy feature (Private softkey) can be configured on the template.

Ideally, you modify templates before registering phones on the network. In this way, you can access customized phone button template options from Cisco Unified Communications Manager during registration.

To modify a phone button template, choose **Device > Device Settings > Phone Button Template** from Cisco Unified Communications Manager Administration. To assign a phone button template to a phone, use the Phone Button Template field in the Cisco Unified Communications Manager Administration Phone Configuration window. Refer to *Cisco Unified Communications Manager Administration Guide* and *Cisco Unified Communications Manager System Guide* for more information.

Modifying a Phone Button Template for Personal Address Book or Fast Dials

You can modify a phone button template to associate a service URL with a line button. Doing so enables users to have single-button access to the PAB and Fast Dials. Before you modify the phone button template, you must configure PAB or Fast Dials as an IP phone service.

To configure PAB or Fast Dial as an IP phone service (if it is not already a service), follow these steps:

Procedure

Step 1 Choose **Device >Device Settings > Phone Services**.

The Find and List IP Phone Services window displays.

Step 2 Click **Add New**. The IP phone services Configuration window displays.

Step 3 Enter the following settings:

- Service Name and ASCII Service Name—Enter **Personal Address Book**.
- Service Description—Enter an optional description of the service.
- Service URL

  For PAB, enter the following URL:

  http://<*Unified CM-server-name*>:8080/ccmpd/login.do?name=#DEVICENAME#&service=pab

  For Fast Dial, enter the following URL:

  http://<*Unified-CM-server-name*>:8080/ccmpd/login.do?name=#DEVICENAME#&service=fd
- Secure Service URL

  For PAB, enter the following URL:

  https://<*Unified CM-server-name*>:8443/ccmpd/login.do?name=#DEVICENAME#&service=pab

  For Fast Dial, enter the following URL:

  https://<*Unified-CM-server-name*>:8443/ccmpd/login.do?name=#DEVICENAME#&service=fd
- Service Category—Select **XML Service**.
- Service Type—Select **Directories**.
- Enable—Select the check box.

Step 4 Click **Save**.

You can add, update, or delete service parameters as needed as described in IP Phone Service Parameter in the *Cisco Unified Communications Manager Administration Guide*.

Note If you change the service URL, remove an IP phone service parameter, or change the name of a phone service parameter for a phone service to which users are subscribed, you must click **Update Subscriptions** to update all currently subscribed users with the changes, or users must resubscribe to the service to rebuild the correct URL.

To modify a phone button template for PAB or Fast Dial, follow these steps:

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose **Device > Device Settings > Phone Button Template**.

Step 2 Click **Find**.

Step 3 Select the phone model.

Step 4 Click **Copy**, enter a name for the new template, and then click **Save**.

The Phone Button Template Configuration window opens.

Step 5 Identify the button you would like to assign, and select **Service URL** from the Features drop-down list box associated with the line.

Step 6 Click **Save** to create a new phone button template using the service URL.

Step 7 Choose **Device > Phone** and open the Phone Configuration window for the phone.

Step 8 Select the new phone button template from the Phone Button Template drop-down list box.

Step 9 Click **Save** to store the change and then click **Apply Config** to implement the change.

The phone user can now access the User Options pages and associate the service with a button on the phone.

For additional information on IP phone services, see the *Cisco Unified Communications Manager Administration Guide,* IP Phone Services Configuration. For additional information on configuring line buttons, see the *Cisco Unified Communications Manager Administration Guide*, Cisco Unified IP Phone Configuration, Configuring Speed-Dial Buttons section.

Configuring Softkey Templates

Using Cisco Unified Communications Manager Administration, you can manage softkeys associated with applications that are supported by the Cisco Unified IP Phone 7906G and 7911G.
Cisco Unified Communications Manager supports two types of softkey templates: standard and nonstandard. Standard softkey templates include Standard User and Standard Feature. An application that supports softkeys can have one or more standard softkey templates associated with it. You can modify a standard softkey template by making a copy of it, giving it a new name, and making updates to that copied softkey template. You can also modify a nonstandard softkey template.

It is recommended that you use the standard softkey template which excludes features already assigned to programmable buttons and limits the feature set to the most commonly used ones. This template reduces the number of softkeys displayed on the phone at one time, eliminating the need for users to press the **more** softkey. For more information, see Modifying Phone Button Templates, page 5-18.

To configure softkey templates, choose **Device > Device Settings > Softkey Template** from Cisco Unified Communications Manager Administration. To assign a softkey template to a phone, use the Softkey Template field in the Cisco Unified Communications Manager Administration Phone Configuration window. Refer to *Cisco Unified Communications Manager Administration Guide* and *Cisco Unified Communications Manager System Guide* for more information.

Note The Cisco IP Phone 7906G and 7911G support all softkeys that are configurable in Cisco Unified Communications Manager Administration, except for the following:

- Hold
- Resume

Setting Up Services

The **Services** button on the Cisco Unified IP Phone gives users access to Cisco Unified IP Phone Services. These services comprise XML applications that enable the display of interactive content with text and graphics on the phone. Examples of services include local movie times, stock quotes, and weather reports.

Before a user can access any service:

- You must use Cisco Unified Communications Manager Administration to configure available services.
- The user must subscribe to services by using Cisco Unified CM User Options. This web-based application provides a graphical user interface (GUI) for limited, end-user configuration of IP Phone applications.

Before you set up services, gather the URLs for the sites that you want to set up and verify that users can access those sites from your corporate IP telephony network.

To set up these services, choose **Feature > Cisco Unified IP Phone Services** from Cisco Unified Communications Manager Administration. Refer to *Cisco Unified Communications Manager Administration Guide* and *Cisco Unified Communications Manager System Guide* for more information.

After you configure these services, verify that your users have access to the Cisco Unified CM User Options web pages, from which they can select and subscribe to configured services. See Appendix A, Providing Information to Usersfor a summary of the information that you must provide to end users.

Cisco Unified IP phones can support up to four HTTP/HTTPS active client connections and up to four HTTP/HTTPS active server connections at one time. A few examples of HTTP/HTTPS services include:

- Extension Mobility
- Directories
- Messages

Adding Users to Cisco Unified Communications Manager

Adding users to Cisco Unified Communications Manager allows you to display and maintain information about users such as their directory information and passwords.

Note You can manage password rules for LDAP directory users by configuring password expiration and syntax in the directory server application that is integrated with Cisco Unified Communications Manager. For more information and a list of supported directory servers, refer to this manual: Installing and Configuring the Cisco Customer Directory Configuration Plugin.

Users added to Cisco Unified Communications Manager can perform these actions:

- Access the corporate directory and other customized directories from a Cisco Unified IP Phone
- Create a personal directory
- Set up speed dial and call forwarding numbers
- Subscribe to services that are accessible from a Cisco Unified IP Phone

You can add users to Cisco Unified Communications Manager using either of these methods:

- To add users individually, choose **User > Add a New User** from Cisco Unified Communications Manager Administration.
- To add users individually, choose **User Management > End User** from Cisco Unified Communications Manager Administration.

  Refer to *Cisco Unified Communications Manager Administration Guide* for more information about adding users. Refer to *Cisco Unified Communications Manager System Guide* for details about user information.
- To add users in batches, use the Bulk Administration Tool (BAT). This method also enables you to set an identical default password for all users.

  Refer to *Bulk Administration Tool User Guide for Cisco Unified Communications Manager* for details.

  Refer to *Cisco Unified Communications Manager Bulk Administration User Guide* for details.

Managing the User Options Web Pages

From the User Options web page, users can customize and control several phone features and settings. For detailed information about the User Options web pages, refer to *Cisco Unified IP Phone 7911G Phone Guide*.

Giving Users Access to the User Options Web Pages

Before a user can access the User Options web pages, you must add the user to the standard Cisco Unified Communications Manager end user group and associate the appropriate phone with the user.

To add the user to the standard Cisco Unified Communications Manager end user group, you must follow these steps:

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose **User Management > User Groups**.

The Find and List Users window displays.

Step 2 Enter the appropriate search criterias and click **Find**.

Step 3 Click on the **Standard CCM End Users** link. The User Group Configuration page for the Standard CCM End Users displays.

Step 4 Click **Add End Users to Group**. The Find and List Users window displays.

Step 5 Use the Find User drop-down list boxes to find the end users that you want to add and click **Find**.

Step 6 A list of end users that matches your search criteria displays.

Step 7 In the list of records that display, click the check box next to the users that you want to add to this user group. If the list comprises multiple pages, use the links at the bottom to see more results.

Note The list of search results does not display end users that already belong to the user group.

Step 8 Click **Add Selected**.

To associate appropriate phones with the user, you must follow these steps:

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose **User Management > End User**.

The Find and List Users window displays.

Step 2 Enter the appropriate search criterias and click **Find**.

Step 3 In the list of records that display, click the link for the user.

Step 4 Click **Device Association.**

The User Device Association window displays.

Step 5 Enter the appropriate search criterias and click **Find**.

Step 6 Choose the device that you want to associate with the end user by checking the box to the left of the device.

Step 7 Click **Save Selected/Changes** to associate the device with the end user.

Make sure to provide end users with the following information about the User Options web pages:

- The URL required to access the application. This URL is:

  http://<*server_name:portnumber***>/ccmuser/**, where *server_name* is the host on which the web server is installed.

- A user ID and default password are needed to access the application.

  These settings correspond to the values you entered when you added the user to Cisco Unified Communications Manager (see Adding Users to Cisco Unified Communications Manager, page 5-21).

For additional information, refer to:

- *Cisco Unified Communications Manager Administration Guide*, User Group Configuration.
- *Cisco Unified Communications Manager Administration Guide*, End User Configuration.
- *Cisco Unified Communications Manager System Guide*, Roles and User Groups.

Specifying Options that Appear on the User Options Web Pages

Most options that are on the User Options web pages appear by default. However, the following options must be set by the system administrator by using Enterprise Parameters Configuration settings in Cisco Unified Communications Manager Administration:

- Show Ring Settings
- Show Line Text Label Settings
- Show Call Forwarding

Note The settings apply to all User Options web pages at your site.

To change the options that appear on the User Options web pages, follow these steps:

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose **System > Enterprise Parameters.**

The Enterprise Parameters Configuration window displays.

Step 2 In the CCMUser Parameters area, specify whether a parameter appears on the User Options web pages by choosing one of these values from the **Parameter Value** drop-down list box for the parameter:

True—Option displays on the User Options web pages (default).

- **False**—Option does not display on the User Options web pages.
- **Show All Settings**—All call forward settings display on the User Options web pages (default).
- **Hide All Settings**—No call forward settings display on the User Options web pages.
- **Show Only Call Forward All**—Only call forward all calls displays on the User Options web pages.

CHAPTER 6

Customizing the Cisco Unified IP Phone

This chapter explains how you customize configuration files, phone ring sounds, background images, and other phone features.

This chapter includes these topics:

- Customizing and Modifying Configuration Files, page 6-1
- Creating Custom Phone Rings, page 6-2
- Creating Custom Background Images, page 6-3

Customizing and Modifying Configuration Files

You can modify configuration files (for example, edit the xml files) and add customized files (for example, custom ring tones, call back tones, phone backgrounds) to the TFTP directory. You can modify files and/or add customized files to the TFTP directory in Cisco Unified Communications Operating System Administration, from the TFTP Server File Upload window. Refer to the *Cisco Unified Communications Operating System Administration Guide* for information on how to upload files to the TFTP folder on a Cisco Unified Communications Manager server.

You can obtain a copy of the Ringlist.xml and List.xml files from the system using the following admin command-line interface (CLI) “file” commands:

- admin:file
  - file list*
  - file view*
  - file search*
  - file get*
  - file dump*
  - file tail*
  - file delete*

Creating Custom Phone Rings

The Cisco Unified IP Phone ships with two default ring types that are implemented in hardware: Chirp1 and Chirp2. Cisco Unified Communications Manager also provides a default set of additional phone ring sounds that are implemented in software as pulse code modulation (PCM) files. The PCM files, along with an XML file (named Ringlist.xml) that describes the ring list options that are available at your site, exist in the TFTP server on each Cisco Unified Communications Manager server.

For more information, see the Custom Phone Rings chapter in the *Cisco Unified Communications Manager Features and Services Guide* and the Software Upgrades chapter in the *Cisco Unified Communications Operating System Administration Guide.*

The following sections describe how you can customize the phone rings that are available at your site by creating PCM files and editing the Ringlist.xml file:

- Ringlist.xml File Format Requirements, page 6-2
- PCM File Requirements for Custom Ring Types, page 6-3
- Configuring a Custom Phone Ring, page 6-3

Ringlist.xml File Format Requirements

The Ringlist.xml file defines an XML object that contains a list of phone ring types. This file can include up to 50 ring types. Each ring type contains a pointer to the PCM file that is used for that ring type and the text that will appear on the Ring Type menu on a Cisco Unified IP Phone for that ring. The Cisco TFTP server for each Cisco Unified Communications Manager contains this file.

The CiscoIPPhoneRinglist XML object uses the following simple tag set to describe the information:

```
<CiscoIPPhoneRinglist>
   <Ring>
      <DisplayName/>
      <FileName/>
   </Ring>
</CiscoIPPhoneRinglist>
```

The following characteristics apply to the definition names. You must include the required DisplayName and FileName for each phone ring type.

- DisplayName defines the name of the custom ring for the associated PCM file that will display on the Ring Type menu of the Cisco Unified IP Phone.
- FileName specifies the name of the PCM file for the custom ring to associate with DisplayName.

Note The DisplayName and FileName fields must not exceed 25 characters.

This example shows a Ringlist.xml file that defines two phone ring types:

```
<CiscoIPPhoneRinglist>
   <Ring>
      <DisplayName>Analog Synth 1</DisplayName>
      <FileName>Analog1.raw</FileName>
   </Ring>
   <Ring>
      <DisplayName>Analog Synth 2</DisplayName>
      <FileName>Analog2.raw</FileName>
   </Ring>
</CiscoIPPhoneRinglist>
```

PCM File Requirements for Custom Ring Types

The PCM files for the rings must meet these requirements for proper playback on Cisco Unified IP Phones:

- Raw PCM (no header)
- 8000 samples per second
- 8 bits per sample
- µLaw compression
- Maximum ring size—16080 samples
- Minimum ring size—240 samples
- Number of samples in the ring is evenly divisible by 240.
- Ring starts and ends at the zero crossing.
- To create PCM files for custom phone rings, you can use any standard audio editing packages that support these file format requirements.

Configuring a Custom Phone Ring

To create custom phone rings for the Cisco Unified IP Phone, follow these steps:

Procedure

Step 1 Create a PCM file for each custom ring (one ring per file). Ensure the PCM files comply with the format guidelines that are listed in the PCM File Requirements for Custom Ring Types, page 6-3.

Step 2 Upload the new PCM files that you created to the Cisco TFTP server for each Cisco Unified Communications Manager in your cluster. For more information, see Software Upgrades in the *Cisco Unified Communications Operating System Administration Guide.*

Step 3 Use an text editor to edit the Ringlist.xml file. See Ringlist.xml File Format Requirements, page 6-2 for information about how to format this file and for a sample Ringlist.xml file.

Step 4 Save your modifications and close the Ringlist.xml file.

Step 5 To cache the new Ringlist.xml file, stop and start the TFTP service by using Cisco Unified Manager Serviceability or disable and re-enable the “Enable Caching of Constant and Bin Files at Startup” TFTP service parameter (located in the Advanced Service Parameters).

Creating Custom Background Images

You can provide users with a choice of background images for the LCD screen on their phones. Users can select a background image by pressing the **Applications Menu** button and choosing **Settings > User Preferences > Background Images** on the phone.

The image choices that users see come from PNG images and an XML file (called List.xml) that are stored on the TFTP server used by the phone. By storing your own PNG files and editing the XML file on the TFTP server, you can designate the background images from which users can choose. In this way, you can provide custom images, such as your company logo.

The following sections describe how you can customize the background images that are available at your site by creating your own PNG files and editing the List.xml file:

- List.xml File Format Requirements, page 6-4.
- PNG File Requirements for Custom Background Images, page 6-4.
- Configuring a Custom Background Image, page 6-5

List.xml File Format Requirements

The List.xml file defines an XML object that contains a list of background images. The List.xml file is stored in the following subdirectory on the TFTP server:

/Desktops/95x34x1

For more information, see the Software Upgrades chapter in the *Cisco Unified Operating System Administration Guide.*

The List.xml file can include up to 50 background images. The images are in the order that they appear in the Background Images menu on the phone. For each image, the List.xml file contains one element type, called ImageItem. The ImageItem element includes these two attributes:

- Image—Uniform resource identifier (URI) that specifies where the phone obtains the thumbnail image that will appear on the Background Images menu on a Phone.
- URI—URI that specifies where the phone obtains the full size image.

The following example shows a List.xml file that defines two images. The required Image and URL attributes must be included for each image. The TFTP URI that is shown in the example is the only supported method for linking to full size and thumbnail images. HTTP URL support is not provided.

List.xml Example

```
<CiscoIPPhoneImageList>
- <!--
 Please Add Images to the end of the list
-->
<ImageItem Image="TFTP:Desktops/95x34x1/TN-Mountain.png"
URL="TFTP:Desktops/95x34x1/Mountain.png" />
<ImageItem Image="TFTP:Desktops/95x34x1/TN-Ocean.png"
URL="TFTP:Desktops/95x34x1/Ocean.png" />
</CiscoIPPhoneImageList>
```

The Cisco Unified IP Phone firmware includes a default background image. This image is not defined in the List.xml file. The default image is always the first image that appears in the Background Images menu on the phone.

PNG File Requirements for Custom Background Images

Each background image requires two PNG files:

- Full size image—Version that appears on the on the phone.
- Thumbnail image—Version that appears on the Background Images screen from which users can select an image. Must be 25% of the size of the full size image.

Tip Many graphics programs provide a feature that will resize a graphic. An easy way to create a thumbnail image is to first create and save the full size image, then use the sizing feature in the graphics program to create a version of that image that is 25% of the original size. Save the thumbnail version using a different name.

The PNG files for background images must meet the following requirements for proper display on the Cisco Unified IP Phone:

- Full size image—95 pixels (width) X 34 pixels (height).
- Thumbnail image—23 pixels (width) X 8 pixels (height).
- Color palette—For best results, set to monochrome (1-bit) when you create a PNG file.

Configuring a Custom Background Image

To configure custom background images for the Cisco Unified IP Phone, follow these steps:

Procedure

Step 1 Create two PNG files for each image (a full size version and a thumbnail version). Ensure the PNG files comply with the format guidelines that are listed in the PNG File Requirements for Custom Background Images, page 6-4.

Step 2 Upload the new PNG files that you created to the following subdirectory in the TFTP server for the Cisco Unified Communications Manager:

/Desktops/95x34x1

Note The file name and subdirectory parameters are case sensitive. Be sure to use the forward slash "/" when you specify the subdirectory path.

To upload the files, choose **Software Upgrades > Upload TFTP Server File** in Cisco IPT Platform Administration. For more information, see the Software Upgrades chapter in *Cisco Unified Communications Operating System Administration Guide*.

Note If the folder does not exist, the folder gets created and the files get uploaded to the folder.

Step 3 You must also copy the customized images and files to the other TFTP servers that the phone may contact to obtain these files.

Note Cisco recommends that you also store backup copies of custom image files in another location. You can use these backup copies if the customized files are overwritten when you upgrade Cisco Unified Communications Manager.

Step 4 Use a text editor to edit the List.xml file. See List.xml File Format Requirements, page 6-4 for the location of this file, formatting requirements, and a sample file.

Step 5 Save your modifications and close the List.xml file.

Note When you upgrade Cisco Unified Communications Manager, a default List.xml file will replace your customized List.xml file. After you customize the List.xml file, make a copy of the file and store it in another location. After upgrading Cisco Unified Communications Manager, replace the default List.xml file with your stored copy.

Step 6 To cache the new List.xml file, stop and start the TFTP service by using Cisco Unified Serviceability or disable and re-enable the Enable Caching of Constant and Bin Files at Startup TFTP service parameter (located in the Advanced Service Parameters).

Custom Background Images for Large Font Locales

Phone background images may not display properly when large font locales such as Chinese, Japanese, and Korean are used. To modify a background image for proper display, follow these guidelines:

Use the following file sizes when creating PNG files for the Japanese locale:

- 95x28 (full size image)
- 23x8 (thumbnail image)

Upload the image files to %TFTPPATH%\Desktops\95x28x1.

Modify or create the List.xml file in the %TFTPPATH%\Desktops\95x28x1 folder to include the following lines, where image.png is the name of your image file:

```
<CiscoIPPhoneImageList>
<ImageItem Image="TFTP:Desktops/95x28x1/image.png"
URL="TFTP:Desktops/95x28x1/image.png" />
</CiscoIPPhoneImageList>
```

For more information, see Creating Custom Background Images, page 6-3 in the Cisco Unified IP Phone 7906G and 7911G for *Cisco Unified Communications Manager Administration Guide*.

Configuring Wideband Codec

If Cisco Unified Communications Manager has been configured to use G.722 (G.722 is enabled by default for the Cisco Unified IP Phone 7970 Series) and if the far endpoint supports G.722, the call can connect using the G.722 codec in place of G.711. The user may notice greater audio sensitivity during the call. Greater sensitivity means improved audio clarity but also means that more background noise can be heard by the far endpoint—noise such as rustling papers or nearby conversations. Even without a wideband handset, some users may prefer the additional sensitivity of G.722. Other users may be distracted by the additional sensitivity of G.722.

Two parameters in Cisco Unified Communications Manager affect whether wideband is supported for this Cisco Unified Communications Manager server and/or a specific phone:

- Advertise G.722 Codec—From Cisco Unified Communications Manager, choose **System > Enterprise Parameters**. The default value of this enterprise parameter is *True*, which means that all Cisco Unified IP Phone Models 7906G, 7911G, 7941G, 7941G-GE, 7961G, 7961G-GE, 7970G,

and 7971G-GE that are registered to this Cisco Unified Communications Manager will advertise G.722 to Cisco Unified Communications Manager. For more information, see *Cisco Unified Communications Manager System Guide*, Cisco Unified IP Phones.

- Advertise G.722 Codec—From Cisco Unified Communications Manager, choose **Device > Phone**. The default value of this product-specific parameter is to use the value specified in the enterprise parameter. If you want to override this on a per-phone basis, choose *Enabled* or *Disabled* in the Advertise G.722 Codec parameter on the Product Specific Configuration area of the Phone Configuration window.

CHAPTER 7

Viewing Model Information, Status, and Statistics on the Cisco Unified IP Phone

This chapter describes how to use the following menus on the Cisco Unified IP Phones 7906G and 7911G to view model information, status messages, network statistics, and firmware information for the phone:

- Model Information screen—Displays hardware and software information about the phone. For more information, see Model Information Screen, page 7-1.
- Status menu—Provides access to screens that display the status messages, network statistics, and firmware versions. For more information, see Status Menu, page 7-2.

You can use the information on these screens to monitor the operation of a phone and to assist with troubleshooting.

You can also obtain much of this information, and obtain other related information, remotely through the phone's web page. For more information, see Monitoring the Cisco Unified IP Phone Remotely.

For more information about troubleshooting the Cisco Unified IP Phone 7906G and 7911G, see Troubleshooting and Maintenance.

This chapter includes these topics:

- Model Information Screen, page 7-1
- Status Menu, page 7-2

Model Information Screen

The Model Information screen displays specific information about the IP phone. To display the Model Information screen, follow these steps:

Procedure

Step 1 Press the **Applications Menu** button.

Step 2 Select **Settings > Model Information.**

Table 7-1 provides a list of Model Information items and a description of each.

Table 7-1 Model Information Settings

Option	Description	To Change
Model Number	Model number of the phone	Display only—Cannot configure
MAC Address	MAC address of the phone	Display only—Cannot configure
Load File	Identifier of the factory-installed load running on the phone	Display only—Cannot configure
Boot Load ID	Identifier of the factory-installed load running on the phone	Display only—Cannot configure
Serial Number	Serial number of the phone	Display only—Cannot configure
MIC	Indicates whether a manufacturing installed certificate (used for the security features) is installed on the phone (Yes) or is not installed on the phone (No).	For information about how to manage the MIC for a phone, refer to Using the Certificate Authority Proxy Function in *Cisco Unified Communications Manager Security Guide*.
LSC	Indicates whether a locally significant certificate (used for the security features) is installed on the phone (Yes) or is not installed on the phone (No)	For information about how to manage the MIC for a phone, refer to Using the Certificate Authority Proxy Function in *Cisco Unified Communications Manager Security Guide*.
Call Control Protocol	Displays the call control protocol for the phone, Skinny Client Control Protocol (SCCP).	See Using Cisco Unified IP Phones with Different Protocols, page 2-10.

Status Menu

The Status menu contains the following options, which provide information about the phone and its operation:

To access the Status menu, follow these steps:

Procedure

Step 1 Press the **Applications Menu** button.

Step 2 Select **Settings > Status Menu.**

Table 7-2 provides a list of Status menu options and a description of each.

Table 7-2 Status Messages on the Cisco Unified IP Phone

Item	Description
Status Messages	Displays the Status Messages screen, which shows a log of important system messages. For more information, see Status Messages Screen, page 7-3.
Network Statistics	Displays the Network Statistics screen, which shows Ethernet traffic statistics. For more information, see Network Statistics Screen, page 7-8.

Table 7-2 Status Messages on the Cisco Unified IP Phone (continued)

Item	Description
Firmware Versions	Displays the Firmware Versions screen, which shows information about the firmware running on the phone. For more information, see Firmware Versions Screen, page 7-11.
802.1X Authentication Status	Displays the time-stamped authentication successes and failures. For more information, see Call Statistics Screen, page 7-12.

Status Messages Screen

The Status Messages screen displays the 10 most recent status messages that the phone has generated. You can access this screen at any time, even if the phone has not finished starting up. Table 7-3 describes the status messages that might appear. This table also includes actions you can take to address errors.

To display the Status Messages screen, follow these steps:

Procedure

Step 1 Press the **Applications Menu** button.

Step 2 Select **Settings**.

Step 3 Select **Status.**

Step 4 Select **Status Messages.**

To remove current status messages, press the **Clear** softkey.

To exit the Status Messages screen, press the **Exit** softkey.

Table 7-3 Status Messages on the Cisco Unified IP Phone 7906G and 7911G

Message	Description	Possible Explanation and Action
BootP server used	The phone obtained its IP address from a BootP server rather than from a DHCP server.	None. This message is informational only.
CFG file not found	The name-based and default configuration file was not found on the TFTP Server.	The configuration file for a phone is created when the phone is added to the Cisco Unified Communications Manager database. If the phone has not been added to the Cisco Unified Communications Manager database, the TFTP server generates a `CFG File Not Found` response. • Phone is not registered with Cisco Unified Communications Manager. You must manually add the phone to Cisco Unified Communications Manager if you are not allowing phones to auto-register. See Adding Phones with Cisco Unified Communications Manager Administration, page 2-9 for details. • If you are using DHCP, verify that the DHCP server is pointing to the correct TFTP server. • If you are using static IP addresses, check configuration of the TFTP server. See Network Configuration Menu, page 4-5 for details about assigning a TFTP server.
CFG TFTP Size Error	The configuration file is too large for file system on the phone.	Power cycle the phone.
Checksum Error	Downloaded software file is corrupted.	Obtain a new copy of the phone firmware and place it in the tftp directory. You should only copy files into this directory when the TFTP server software is shut down, otherwise the files may be corrupted.
CTL installed	The CTL file is installed in the phone.	None. This message is informational only. The CTL file was not installed previously. For more information about the CTL file, refer to *Cisco Unified Communications Manager Security Guide*.
CTL and ITL installed	The CTL and ITL files are installed on the phone.	None. This message is informational only. Phone does not have prior installation of either CTL or ITL file. For more information about the CTL file, refer to *Cisco Unified Communications Manager Security Guide*.
DHCP timeout	DHCP server did not respond.	• Network is busy—The errors should resolve themselves when the network load reduces. • No network connectivity between the DHCP server and the phone—Verify the network connections. • DHCP server is down—Check configuration of DHCP server. • Errors persist—Consider assigning a static IP address. See Network Configuration Menu, page 4-5 for details on assigning a static IP address.

Table 7-3 Status Messages on the Cisco Unified IP Phone 7906G and 7911G (continued)

Message	Description	Possible Explanation and Action
Disabled	802.1X Authentication is disabled on the phone.	You can enable 802.1X using the **Settings > Security Configuration > 802.1X Authentication** option on the phone. For more information, see 802.1X Authentication and Status, page 4-37.
DNS timeout	DNS server did not respond.	• Network is busy—The errors should resolve themselves when the network load reduces. • No network connectivity between the DNS server and the phone—Verify the network connections. • DNS server is down—Check configuration of DNS server.
DNS unknown host	DNS could not resolve the name of the TFTP server or Cisco Unified Communications Manager.	• Verify that the host names of the TFTP server or Cisco Unified Communications Manager are configured properly in DNS. • Consider using IP addresses rather than host names.
Duplicate IP	Another device is using the IP address assigned to the phone.	• If the phone has a static IP address, verify that you have not assigned a duplicate IP address. See Network Configuration Menu, page 4-5 for details. • If you are using DHCP, check the DHCP server configuration.
Erasing CTL and ITL files	Erasing CTL or ITL file.	None. This message is informational only. For more information about the CTL and ITL files, refer to *Cisco Unified Communications Manager Security Guide*.
Error update locale	One or more localization files could not be found in the tftp directory or were not valid. The locale was not changed.	From Cisco Unified Operating System Administration, check that the following files are located within subdirectories in the TFTP File Management: • Located in subdirectory with same name as network locale: – tones.xml • Located in subdirectory with same name as user locale: – glyphs.xml – dictionary.xml – kate.xml
Failed	The phone attempted an 802.1X transaction but authentication failed.	Authentication typically fails for of one of the following reasons: • No shared secret is configured in the phone or authentication server. • The shared secret configured in the phone and the authentication server do not match. • Phone has not been configured in the authentication server.

Table 7-3 Status Messages on the Cisco Unified IP Phone 7906G and 7911G (continued)

Message	Description	Possible Explanation and Action
File auth error	An error occurred when the phone tried to validate the signature of a signed file. This message includes the name of the file that failed.	• The file is corrupted. If the file is a phone configuration file, delete the phone from the Cisco Unified Communications Manager database using Cisco Unified Communications Manager Administration. Then add the phone back to the Cisco Unified Communications Manager database using Cisco Unified Communications Manager Administration. • There is a problem with the CTL file and the key for the server from which files are obtained is bad. In this case, run the CTL client and update the CTL file, making sure that the proper TFTP servers are included in this file.
File not found	The phone cannot locate, on the TFTP server, the phone load file that is specified in the phone configuration file.	From Cisco Unified Operating System Administration, make sure that the phone load file is on the TFTP server, and that the entry in the configuration file is correct.
IP address released	The phone has been configured to release its IP address.	The phone remains idle until it is power cycled or you reset the DHCP address. See Network Configuration Menu, page 4-5 for details.
ITL installed	The ITL file is installed in the phone.	None. This message is informational only. The ITL file was not installed previously. For more information about the CTL file, refer to *Cisco Unified Communications Manager Security Guide*.
Load Auth Failed	The phone could not load a configuration file.	Check that: • A good version of the configuration file exists on the applicable server. • The phone load being downloaded has not been altered or renamed. • Phone load type is compatible; for example, you cannot place a DEV load configuration file on a REL-signed phone.
Load ID incorrect	Load ID of the software file is of the wrong type.	Check the load ID assigned to the phone (from Cisco Unified Communications Manager, choose **Device > Phone**). Verify that the load ID is entered correctly.
Load rejected HC	The application that was downloaded is not compatible with the phone's hardware.	Occurs if you were attempting to install a version of software on this phone that did not support hardware changes on this newer phone. Check the load ID assigned to the phone (from Cisco Unified Communications Manager, choose **Device > Phone**). Re-enter the load displayed on the phone. See Firmware Versions Screen, page 7-11 to verify the phone setting.
Load Server is invalid	Indicates an invalid TFTP server IP address or name in the Load Server option.	The Load Server setting is not valid. The Load Server specifies a TFTP server IP address or name from which the phone firmware can be retrieved for upgrades on the phones. Check the Load Server entry (from Cisco Unified Communications Manager Administration choose **Device > Phone**).

Table 7-3 Status Messages on the Cisco Unified IP Phone 7906G and 7911G (continued)

Message	Description	Possible Explanation and Action
No default router	DHCP or static configuration did not specify a default router.	• If the phone has a static IP address, verify that the default router has been configured. See Network Configuration Menu, page 4-5 for details. • If you are using DHCP, the DHCP server has not provided a default router. Check the DHCP server configuration.
No DNS server IP	A name was specified but DHCP or static IP configuration did not specify a DNS server address.	• If the phone has a static IP address, verify that the DNS server has been configured. See Network Configuration Menu, page 4-5 for details. • If you are using DHCP, the DHCP server has not provided a DNS server. Check the DHCP server configuration.
No Trust List installed	The CTL file or the ITL file is not installed on the phone.	The Trust List is not configured on the Cisco Unified Communications Manager, which does not support security by default. For more information about CTL and ITL files, see the *Cisco Unified Communications Manager Security Guide*.
Programming Error	The phone failed during programming.	Attempt to resolve this error by power cycling the phone. If the problem persists, contact Cisco technical support for additional assistance.
Successful – MD5	The phone attempted an 802.1X transaction and authentication achieved.	The phone achieved 802.1X authentication.
TFTP access error	TFTP server is pointing to a directory that does not exist.	• If you are using DHCP, verify that the DHCP server is pointing to the correct TFTP server. • If you are using static IP addresses, check configuration of TFTP server. See Network Configuration Menu, page 4-5 for details on assigning a TFTP server.
TFTP Error	The phone does not recognize an error code provided by the TFTP server.	Contact the Cisco TAC.
TFTP file not found	The requested load file (.bin) was not found in the tftp directory.	Check the load ID assigned to the phone (from Cisco Unified Communications Manager, choose **Device > Phone**). Verify that the tftp directory contains a .bin file with this load ID as the name.
TFTP timeout	TFTP server did not respond.	• Network is busy—The errors should resolve themselves when the network load reduces. • No network connectivity between the TFTP server and the phone—Verify the network connections. • TFTP server is down—Check configuration of TFTP server.
Timed out	Supplicant attempted 802.1X transaction but timed out to due the absence of an authenticator.	Authentication typically times out if 802.1X is not configured on the switch.

Table 7-3 Status Messages on the Cisco Unified IP Phone 7906G and 7911G (continued)

Message	Description	Possible Explanation and Action
Trust List updated	The CTL file, the ITL file, or both files are updated.	None. This message is informational only. For more information about the Trust List, refer to *Cisco Unified Communications Manager Security Guide*.
Trust List update failed	Updating CTL file and ITL failed.	The CTL and ITL files are installed on the phone, and it failed to update the new files. Possible reasons for failure: • Network failure • TFTP server was down • The new security token used to sign CTL file and the TFTP certificate used to sign ITL file are introduced, but are not available in the current CTL and ITL files in the phone • Internal phone failure Possible solutions: • Check the network connectivity • Check if the TFTP server is active and functioning normally • If the TVS server is supported on Cisco Unified Communications Manager, check if the TVS server is active and functioning normally • Verify if the security token and the TFTP server are valid • Manually delete the CTL and ITL files if all the above solutions fail, and reset the phone.
Version error	The name of the phone load file is incorrect.	Make sure that the phone load file has the correct name.
XmlDefault corresponding to the phone device name	Name of the configuration file.	None. This is an informational message indicating the name of the configuration file for the phone.

Network Statistics Screen

The Network Statistics screen displays information about the phone and network performance.

To display the Network Statistics screen, follow these steps:

Procedure

Step 1 Press the **Applications Menu** button.

Step 2 Select **Settings.**

Step 3 Select **Status.**

Step 4 Select **Network Statistics.**

To reset the Rx Frames, Tx Frames, and Rx Broadcasts statistics to 0, press the **Clear** softkey.

Table 7-4 provides a list of Network Statistics items and a description of each.

Table 7-4 Network Statistics Screen

Item	Description
Rx Frames	Number of packets received by the phone
Tx Frames	Number of packets sent by the phone
Rx Broadcasts	Number of broadcast packets received by the phone
One of the following values: Initialized TCP-timeout CM-closed-TCP TCP-Bad-ACK CM-reset-TCP CM-aborted-TCP CM-NAKed KeepaliveTO Failback Phone-Keypad Phone-Re-IP Reset-Reset Reset-Restart Phone-Reg-Rej Load Rejected HC CM-ICMP-Unreach Phone-Abort	Cause of the last reset of the phone
Elapsed Time	Amount of time that has elapsed since the phone connected to Cisco Unified Communications Manager
Port 1	Link state and connection of the Network port
Port 2 (applies to 7911G only)	Link state and connection of the PC port (for example, `Auto 100 Mb Full-Duplex` means that the PC port is in a link up state and has auto-negotiated a full-duplex, 100-Mbps connection)

Table 7-4 Network Statistics Screen (continued)

Item	Description
IPv4	Information on the DHCP status. This includes the following states: • CDP BOUND • CDP INIT • DHCP BOUND • DHCP DISABLED • DHCP INIT • DHCP INVALID • DHCP REBINDING • DHCP REBOOT • DHCP RENEWING • DHCP REQUESTING • DHCP RESYNC • DHCP UNRECOGNIZED • DHCP WAITING COLDBOOT TIMEOUT • SET DHCP COLDBOOT • SET DHCP DISABLED • DISABLED DUPLICATE IP • SET DHCP FAST

Table 7-4 Network Statistics Screen (continued)

Item	Description
IPv6	Information on the DHCPv6 status.This includes the following states: • DHCP6 BOUND; • DHCP6 DISABLED • DHCP6 RENEW • DHCP6 REBIND • DHCP6 INIT • DHCP6 SOLICIT • DHCP6 REQUEST • DHCP6 RELEASING • DHCP6 RELEASED • DHCP6 DISABLING • DHCP6 DECLINING • DHCP6 DECLINED • DHCP6 INFOREQ • DHCP6 INFOREQ DONE • DHCP6 INVALID • DHCP6 DECLINED DUPLICATE IP • DHCP6 WAITING COLDBOOT TIMEOUT • DHCP6 TIMEOUT USING RESTORED VAL • DHCP6 TIMEOUT. CANNOT RESTORE • STACK TURNED OFF

Firmware Versions Screen

The Firmware Versions screen displays information about the firmware version running on the phone.

To display the Firmware Version screen, follow these steps:

Procedure

Step 1 Press the **Applications Menu** button.

Step 2 Select **Settings. > Status**.

Step 3 Select **Firmware Versions.**

To exit the Firmware Version screen, press the **Exit** softkey.

Table 7-5 provides a list of Firmware Version items and a description of each.

Table 7-5 Firmware Version Information

Item	Description
Load File	Load file running on the phone
App Load ID	Identifies the JAR file running on the phone
JVM Load ID	Identifies the Java Virtual Machine (JVM) running on the phone
OS Load ID	Identifies the operating system running on the phone
Boot Load ID	Identifies the factory-installed load running on the phone
DSP Load ID	Identifies the DSP load file running on the phone.

Call Statistics Screen

You can access the Call Statistics screen on the phone to display counters, statistics, and voice-quality metrics. After a call, you can view the call information captured during the last call by displaying the Call Statistics screen.

Note You can remotely view the call statistics information by using a web browser to access the Streaming Statistics web page. This web page contains additional RTCP statistics that are not available on the phone. For more information about remote monitoring, see Streaming Statistics, page 8-11.

A single call can have multiple voice streams, but data is captured for only the last voice stream. A voice stream is a packet stream between two endpoints. If one endpoint is put on hold, the voice stream stops even though the call is still connected. When the call resumes, a new voice packet stream begins, and the new call data overwrites the former call data.

To display the Call Statistics screen for information about the last voice stream, follow these steps:

Procedure

Step 1 Press the **Settings** button.

Step 2 Select **Status**.

Step 3 Select **Call Statistics**.

The Call Statistics screen displays these items:

Table 7-6 Call Statistics Items

Item	Description
Rcvr Codec	Type of voice stream received (RTP streaming audio from codec): G.729, G.728/iLBC, G.711 u-law, G.711 A-law, or Lin16k.
Sender Codec	Type of voice stream transmitted (RTP streaming audio from codec): G.729, G.728/iLBC, G.711 u-law, G.711 A-law, or Lin16k.

Table 7-6 Call Statistics Items (continued)

Item	Description
Rcvr Size	Size of voice packets, in milliseconds, in the receiving voice stream (RTP streaming audio).
Sender Size	Size of voice packets, in milliseconds, in the transmitting voice stream.
Rcvr Packets	Number of RTP voice packets received since voice stream was opened. **Note** This number is not necessarily identical to the number of RTP voice packets received since the call began because the call might have been placed on hold.
Sender Packets	Number of RTP voice packets transmitted since voice stream was opened. **Note** This number is not necessarily identical to the number of RTP voice packets transmitted since the call began because the call might have been placed on hold.
Avg Jitter	Estimated average RTP packet jitter (dynamic delay that a packet encounters when going through the network) observed since the receiving voice stream was opened.
Max Jitter	Maximum jitter observed since the receiving voice stream was opened.
Rcvr Discarded	Number of RTP packets in the receiving voice stream that have been discarded (bad packets, too late, and so on). **Note** The phone will discard payload type 19 comfort noise packets that are generated by Cisco Gateways, which will increment this counter.
Rcvr Lost Packets	Missing RTP packets (lost in transit).
Voice Quality Metrics	
MOS LQK	Score that is an objective estimate of the mean opinion score (MOS) for listening quality (LQK) that rates from 5 (excellent) to 1 (bad). This score is based on audible concealment events due to frame loss in the preceding 8-second interval of the voice stream. For more information, see Monitoring the Voice Quality of Calls, page 9-14. **Note** The MOS LQK score can vary based on the type of codec that the Cisco Unified IP Phone uses.
Avg MOS LQK	Average MOS LQK score observed for the entire voice stream.
Min MOS LQK	Lowest MOS LQK score observed from start of the voice stream.

Table 7-6 Call Statistics Items (continued)

Item	Description
Max MOS LQK	Baseline or highest MOS LQK score observed from start of the voice stream. These codecs provide the following maximum MOS LQK score under normal conditions with no frame loss: • G.711 gives 4.5 • G.722 gives 4.5 • G.728/iLBC gives 3.9 • G.729 A/AB gives 3.8
MOS LQK Version	Version of the Cisco proprietary algorithm used to calculate MOS LQK scores.
Cumulative Conceal Ratio	Total number of concealment frames divided by total number of speech frames received from start of the voice stream.
Interval Conceal Ratio	Ratio of concealment frames to speech frames in preceding 3-second interval of active speech. If using voice activity detection (VAD), a longer interval might be required to accumulate 3 seconds of active speech.
Max Conceal Ratio	Highest interval concealment ratio from start of the voice stream.
Conceal Secs	Number of seconds that have concealment events (lost frames) from the start of the voice stream (includes severely concealed seconds).
Severely Conceal Secs	Number of seconds that have more than 5 percent concealment events (lost frames) from the start of the voice stream.
Network Protocol	Identifies the current Network Protocol—IPv4.
Latency[1]	Estimate of the network latency, expressed in milliseconds. Represents a running average of the round-trip delay, measured when RTCP receiver report blocks are received.

1. When the RTP Control Protocol is disabled, no data generates for this field and thus displays as 0.

Using Test Tone

The Cisco Unified IP Phone supports a "test tone," which allows you to troubleshoot echo on a call as well as to test low volume levels.

To use a test tone, you must:

- Enable the tone generator
- Create a test tone

To enable the tone generator, follow these steps:

Procedure

Step 1 Verify that the phone is unlocked.

When options are inaccessible for modification, a *locked* padlock icon appears on the configuration menus. When options are unlocked and accessible for modification, an *unlocked* padlock icon appears on these menus.

To unlock or lock options on the Settings menu, press ****#** on the phone keypad. This action either locks or unlocks the options, depending on the previous state.

Note If a Settings Menu password has been provisioned, SIP phones present an "Enter password" prompt after you enter **#.

Make sure to lock options after you have made your changes.

Caution Do not press ****#** to unlock options and then immediately press ****#** again to lock options. The phone will interpret this sequence as ****#****, which will reset the phone. To lock options after unlocking them, wait at least 10 seconds before you press ****#** again.

Step 2 While offhook, press the Help button twice to invoke the Call Statistics screen, or press **Settings > Status > Call Statistics** to invoke the Call Statistics screen. Look for the Tone softkey.

When the Tone softkey is visible, the softkey remains enabled for as long as this Cisco Unified IP phone is registered with Cisco Unified Communications Manager.

You can proceed to the Procedure for creating a test tone.

Step 3 If the Tone softkey is not present, exit the Call Statistics screen and enter the Setting Menu. Press ****3** on the phone keypad to enable the Tone softkey. (Pressing ****3** toggles the Tone softkey.)

Note If you press ****# **3** consecutively, with no pause, you will inadvertently reset the phone because of the ****#**** sequence.

While offhook, press the Help button twice to invoke the Call Statistics screen, or press **Settings > Status > Call Statistics** to invoke the Call Statistics screen. Verify that the Tone softkey is present.

When the Tone softkey is visible, the softkey remains enabled for as long as this Cisco Unified IP Phone is registered with Cisco Unified Communications Manager.

To create a test tone, follow these steps:

Procedure

Note When measuring echo, make sure you first set the input and output levels to 0 dB gain/attenuation on the trunk. This is set for the gateway (in Cisco Unified Communications Manager for MGCP) or under IOS CLI for H.323 or SIP.

Step 1 Place a call.

Step 2 Once the call is established, press the Help button twice, or press **Settings > Status > Call Statistics**.

The Call Statistics screen and Tone softkey should appear.

Step 3 Press the **Tone** softkey.

The phone generates a 1004 Hz tone at -15 dBm. For a good network connection, the tone sounds at the call destination only. For a bad network connection, the phone generating the tone may receive echo from the destination phone.

Step 4 To stop the tone, end the call.

For information on interpreting the results of test tone for volume and echo, refer to the following document:

http://www.cisco.com/en/US/docs/ios/solutions_docs/voip_solutions/EA_ISD.html

CHAPTER 8

Monitoring the Cisco Unified IP Phone Remotely

Each Cisco Unified IP Phone has a web page from which you can view a variety of information about the phone, including:

- Device information
- Network configuration information
- Network statistics
- Device logs
- Streaming statistics

Note The Cisco Unified IP Phone does not support web access on its IPv6 address.

This chapter describes the information that you can obtain from the phone's web page. You can use this information to remotely monitor the operation of a phone and to assist with troubleshooting.

You can also obtain much of this information directly from a phone. For more information, see Viewing Model Information, Status, and Statistics on the Cisco Unified IP Phone.

For more information about troubleshooting the Cisco Unified IP Phone 7906G and 7911G, see Troubleshooting and Maintenance.

This chapter includes these topics:

- Accessing the Web Page for a Phone, page 8-1
- Disabling and Enabling Web Page Access, page 8-3
- Device Information, page 8-3
- Network Configuration, page 8-4
- Network Statistics, page 8-8
- Device Logs, page 8-11
- Streaming Statistics, page 8-11

Accessing the Web Page for a Phone

To access the web page for a Cisco Unified IP Phone, perform the following these steps.

Note If you cannot access the web page, it may be disabled. See Disabling and Enabling Web Page Access, page 8-3 for more information.

Procedure

Step 1 Obtain the IP address of the Cisco Unified IP Phone using one of these methods:

- Search for the phone in Cisco Unified Communications Manager by choosing **Device > Phone**. Phones registered with Cisco Unified Communications Manager display the IP address at the top of the Phone Configuration window.
- On the phone, press the **Applications Menu** button, choose **Network Configuration**, and then scroll to the IP Address option.

Step 2 Open a web browser and enter the following URL, where *IP_address* is the IP address of the Cisco Unified IP Phone:

http://*IP_address* or https://*IP_address* (depending on the protocol supported by the Cisco Unified IP Phone)

The web page for Cisco Unified IP Phone includes these hyperlinks:

- **Device Information**—Displays device settings and related information for the phone. For more information, see Device Information, page 8-3.
- **Network Configuration**—Displays network configuration information and information about other phone settings. For more information, see Network Configuration, page 8-4.
- **Network Statistics**—Includes the following hyperlinks, which provide information about network traffic:
  - **Ethernet Information**—Displays information about Ethernet traffic. For more information, see Network Statistics, page 8-8.
  - **Access**—Displays information about network traffic to and from the PC port on the phone. For more information, see Network Statistics, page 8-8.
  - **Network**—Displays information about network traffic to and from the network port on the phone. For more information, see Network Statistics, page 8-8.
- **Device Logs**—Includes the following hyperlinks, which provide information that you can use for troubleshooting:
  - **Console Logs**—Includes hyperlinks to individual log files. For more information, see Device Logs, page 8-11.
  - **Core Dumps**—Includes hyperlinks to individual dump files.
  - **Status Messages**—Displays up to the 10 most recent status messages that the phone has generated since it was last powered up. For more information, see Device Logs, page 8-11.
  - **Debug Display**—Displays messages that might be useful to the Cisco TAC if you require assistance with troubleshooting. For more information, see Device Logs, page 8-11.
- **Streaming Statistics**—Includes the **Stream 1**, **Stream 2**, and **Stream 3** hyperlinks, which display a variety of streaming statistics. For more information, see Streaming Statistics, page 8-11.

Disabling and Enabling Web Page Access

For security purposes, you may choose to prevent access to the web pages for a phone. If you do so, you will prevent access to the web pages that are described in this chapter and to the phone's User Options web pages.

You can enable or disable access to the web pages for an individual phone, a group of phones, or to all phones in the system.

To enable or disable access to the web pages for all phones on the system, choose **System > Enterprise Parameters** and select **Enabled** or **Disabled** from the Web Access drop-down menu.

To enable or disable access to the web pages for a group of phones, choose **Device > Device Settings > Common Phone Profile** to create a new phone profile or to update an existing phone profile, select **Enabled** or **Disabled** from the Web Access drop-down menu and select the common phone profile when you configure your phone.

To enable or disable access to the web pages for a phone, follow these steps from Cisco Unified Communications Manager Administration:

Procedure

Step 1 Choose **Device > Phone**.

Step 2 Specify the criteria to find the phone and click **Find**, or click **Find** to display a list of all phones.

Step 3 Click the device name to open the Phone Configuration window for the device.

Step 4 From the Web Access drop-down list box, choose **Disabled** if you want to disable your phone and choose Enabled if you want to enable your phone.

Step 5 Click **Update**.

Note Some features, such as Cisco Quality Report Tool, do not function properly without access to the phone web pages. Disabling web access also affects any serviceability application that relies on web access, such as CiscoWorks.

Configuring the Cisco Unified IP Phone to use HTTP/HTTPS Protocols

The Cisco Unified IP Phone can be configured to use:

- The HTTPS protocol only
- The HTTP or HTTPS protocols

If your Cisco Unified IP Phone is configured to use the HTTP or HTTPS protocols (the second case above), use http://IP_Address or https://IP_Address for the phone's web access.

Device Information

The Device Information area on a phone's web page displays device settings and related information for the phone. Table 8-1 describes these items.

To display the Device Information area, access the web page for the phone as described in the Accessing the Web Page for a Phone, page 8-1, and then click the **Device Information** hyperlink.

Table 8-1 Device Information Area Items

Item	Description
MAC Address	Media Access Control (MAC) address of the phone
Host Name	Unique, fixed name that is automatically assigned to the phone based on its MAC address
Phone DN	Directory number assigned to the phone
App Load ID	Identifier of the firmware running on the phone
Boot Load ID	Identifier of the factory-installed load running on the phone
Version	Version of the firmware running on the phone
Hardware Revision	Revision value of the phone hardware
Serial Number	Serial number of the phone
Model Number	Model number of the phone
Message Waiting	Indicates if there is a voice message waiting on any line for this phone
UDI	Displays the following Cisco Unique Device Identifier (UDI) information about the phone: • Device Type—Indicates hardware type. For example, phone displays for all phone models • Device Description—Displays the name of the phone associated with the indicated model type • Product Identifier—Specifies the phone model • Version Identifier—Represents the hardware version of the phone The Version Identifier field might display blank if using an older model Cisco Unified IP Phone because the hardware does not provide this information. • Serial Number—Displays the phone's unique serial number
Time	Time obtained from the Date/Time Group in Cisco Unified Communications Manager to which the phone belongs
Time Zone	Timezone obtained from the Date/Time Group in Cisco Unified Communications Manager to which the phone belongs
Date	Date obtained from the Date/Time Group in Cisco Unified Communications Manager to which the phone belongs

Network Configuration

The Network Configuration area on a phone's web page displays network configuration information and information about other phone settings. Table 8-2 describes these items.

You can view and set many of these items from the Network Configuration Menu and the Device Configuration Menu on the Cisco Unified IP Phone. For more information, see Configuring Features, Templates, Services, and Users

To display the Network Configuration area, access the web page for the phone as described in the Accessing the Web Page for a Phone, page 8-1, and then click the **Network Configuration** hyperlink.

Table 8-2 Network Configuration Area Items

Item	Description
DHCP Server	IP address of the Dynamic Host Configuration Protocol (DHCP) server from which the phone obtains its IP address.
BOOTP Server	Indicates whether the phone obtains its configuration from a Bootstrap Protocol (BootP) server.
MAC Address	Media Access Control (MAC) address of the phone.
Host Name	Host name that the DHCP server assigned to the phone.
Domain Name	Name of the Domain Name System (DNS) domain in which the phone resides.
IP Address	Internet Protocol (IP) address of the phone.
Subnet Mask	Subnet mask used by the phone.
TFTP Server 1	Primary Trivial File Transfer Protocol (TFTP) server used by the phone.
Default Router 1–5	Default router used by the phone (Default Router 1) and optional backup routers (Default Router 2–5.
DNS Server 1–5	Primary Domain Name System (DNS) server (DNS Server 1) and optional backup DNS servers (DNS Server 2–5) used by the phone.
Operational VLAN ID	Auxiliary Virtual Local Area Network (VLAN) configured on a Cisco Catalyst switch in which the phone is a member.
Admin. VLAN ID	Auxiliary VLAN in which the phone is a member.
Unified CM1–5	Host names or IP addresses, in prioritized order, of the Cisco Unified Communications Manager servers with which the phone can register. An item can also show the IP address of an SRST router that is capable of providing limited Cisco Unified Communications Manager functionality, if such a router is available. For an available server, an item will show the Cisco Unified Communications Manager server IP address and one of the following states: • Active—Cisco Unified Communications Manager server from which the phone is currently receiving call-processing services. • Standby—Cisco Unified Communications Manager server to which the phone switches if the current server becomes unavailable. • Blank—No current connection to this Cisco Unified Communications Manager server. An option may also include the Survivable Remote Site Telephony (SRST) designation, which indicates an SRST router capable of providing Cisco Unified Communications Manager functionality with a limited feature set. This router assumes control of call processing if all other Cisco Unified Communications Manager servers become unreachable. The SRST Cisco Unified Communications Manager always appears last in the list of servers, even if it is active. You configure the SRST router address in the Device Pool section in Cisco Unified Communications Manager Configuration window.
Information URL	URL of the help text that appears on the phone.
Directories URL	URL of the server from which the phone obtains directory information.
Messages URL	URL of the server from which the phone obtains message services.
Services URL	URL of the server from which the phone obtains Cisco Unified IP Phone services.

Table 8-2 Network Configuration Area Items (continued)

Item	Description
DHCP Enabled	Indicates whether DHCP is being used by the phone.
DHCP Address Released	Indicates the setting of the DHCP Address Released option on the phone's Network Configuration menu.
Alternate TFTP	Indicates whether the phone is using an alternative TFTP server.
Idle URL	URL that the phone displays when the phone has not been used for the time specified by Idle URL Time, and no menu is open.
Idle URL Time	Number of seconds that the phone has not been used and no menu is open before the XML service specified by Idle URL is activated.
Proxy Server URL	URL of proxy server, which makes HTTP requests to non-local host addresses on behalf of the phone HTTP client and provides responses from the non-local host to the phone HTTP client.
Authentication URL	URL that the phone uses to validate requests made to the phone web server.
SW Port Configuration	Speed and duplex of the switch port, where: • A—Auto Negotiate • 10H—10-BaseT/half duplex • 10F—10-BaseT/full duplex • 100H—100-BaseT/half duplex • 100F—100-BaseT/full duplex • No Link—No connection to the switch port
PC Port Configuration (applies to 7911G only)	Speed and duplex of the switch port, where: • A—Auto Negotiate • 10H—10-BaseT/half duplex • 10F—10-BaseT/full duplex • 100H—100-BaseT/half duplex • 100F—100-BaseT/full duplex • No Link—No connection to the PC port
TFTP Server 2	Backup TFTP server that the phone uses if the primary TFTP server is unavailable.
User Locale	User locale associated with the phone user. Identifies a set of detailed information to support users, including language, font, date and time formatting, and alphanumeric keyboard text information.
Network Locale	Network locale associated with the phone user. Identifies a set of detailed information to support the phone in a specific location, including definitions of the tones and cadences used by the phone.
User Locale Version	Version of the user locale loaded on the phone.
Network Locale Version	Version of the network locale loaded on the phone.
PC Port Disabled (applies to 7911G only)	Indicates whether the PC port on the phone is enabled or disabled.
Speaker Enabled	Indicates whether the speakerphone is enabled on the phone.

Table 8-2 Network Configuration Area Items (continued)

Item	Description
Group Listen	Enables both the handset and speaker to be active at the same time, so that one user can talk into the handset while other users listen over the speaker.
GARP Enabled	Indicates whether the phone learns MAC addresses from Gratuitous ARP responses.
Voice VLAN Enabled (applies to 7911G only)	Indicates whether the phone allows a device attached to the PC port to access the Voice VLAN.
Auto Line Select Enabled	Indicates whether the phone shifts the call focus to incoming calls on all lines.
DSCP for Call Control	DSCP IP classification for call control signaling.
DSCP for Configuration	DSCP IP classification for any phone configuration transfer.
DSCP for Services	DSCP IP classification for phone-based services.
Security Mode	Displays the security mode that is set for the phone.
Web Access Enabled	Indicates whether web access is enabled (Yes) or disabled (No) for the phone.
Span to PC Port (applies to 7911G only)	Indicates whether the phone will forward packets transmitted and received on the network port to the access port.
PC VLAN (applies to 7911G only)	VLAN used to identify and remove 802.1P/Q tags from packets sent to the PC.
CDP: PC Port (applies to 7911G only)	Indicates whether CDP is enabled on the PC port (default is enabled).
LLDP: PC Port	Indicates whether Link Layer Discovery Protocol (LLDP) is enabled on the PC port.
LLDP-MED: SW Port	Indicates whether Link Layer Discovery Protocol Media Endpoint Discovery (LLDP-MED) is enabled on the switch port.
LLDP Power Priority	Advertises the phone's power priority to the switch, enabling the switch to appropriately provide power to the phones. Settings include: • Unknown—default • Low • High • Critical
LLDP Asset ID	Identifies the asset ID assigned to the phone for inventory management.
LLDP Power Priority	Advertises the phone's power priority to the switch, enabling the switch to appropriately provide power to the phones. Settings include: • Unknown—default • Low • High • Critical
LLDP Asset ID	Identifies the asset ID assigned to the phone for inventory management.

Table 8-2 Network Configuration Area Items (continued)

Item	Description
LLDP Power Priority	Advertises the phone's power priority to the switch, enabling the switch to appropriately provide power to the phones. Settings include: • Unknown—default • Low • High • Critical
LLDP Asset ID	Identifies the asset ID assigned to the phone for inventory management.
LLDP Power Priority	Advertises the phone's power priority to the switch, enabling the switch to appropriately provide power to the phones. Settings include: • Unknown—default • Low • High • Critical
LLDP Asset ID	Identifies the asset ID assigned to the phone for inventory management.
LLDP Power Priority	Advertises the phone's power priority to the switch, enabling the switch to appropriately provide power to the phones. Settings include: • Unknown—default • Low • High • Critical
LLDP Power Priority	Advertises the phone's power priority to the switch, enabling the switch to appropriately provide power to the phones. Settings include: • Unknown—default • Low • High • Critical
LLDP Asset ID	Identifies the asset ID assigned to the phone for inventory management.
LLDP AgeoutsTotal	Total number of LLDP frames that have been time out in cache

Network Statistics

These network statistics areas on a phone's web page provide information about network traffic on the phone:

- Ethernet Information area—Displays information about Ethernet traffic. Table 8-3 describes the items in this area.
- Access area—Displays information about network traffic to and from the PC port on the phone. Table 8-4 describes the items in this area.

- Network area—Displays information about network traffic to and from the network port on the phone. Table 8-4 describes the items in this area.

To display a network statistics area, access the web page for the phone as described in the Accessing the Web Page for a Phone, page 8-1, and then click the **Ethernet Information**, the **Access**, and or the **Network** hyperlink.

Table 8-3 Ethernet Information Area Items

Item	Description
Tx Frames	Total number of packets transmitted by the phone
Tx broadcast	Total number of broadcast packets transmitted by the phone
Tx multicast	Total number of multicast packets transmitted by the phone
Tx unicast	Total number of unicast packets transmitted by the phone
Rx Frames	Total number of packets received by the phone
Rx broadcast	Total number of broadcast packets received by the phone
Rx multicast	Total number of multicast packets received by the phone
Rx unicast	Total number of unicast packets received by the phone
RxPacketNoDes	Total number of shed packets caused by no direct memory access (DMA) descriptor

Table 8-4 Access Area and Network Area Items

Item	Description
Rx totalPkt	Total number of packets received by the phone
Rx crcErr	Total number of packets received with CRC failed
Rx alignErr	Total number of packets received between 64 and 1522 bytes in length that have a bad FCS
Rx multicast	Total number of multicast packets received by the phone
Rx broadcast	Total number of broadcast packets received by the phone
Rx unicast	Total number of unicast packets received by the phone
Rx shortErr	Total number of frame check sequence (FCS) error packets or Align error packets received that are less than 64 bytes in size
Rx shortGood	Total number of good packets received that are less than 64 bytes size
Rx longGood	Total number of good packets received that are greater than 1522 bytes in size
Rx longErr	Total number of FCS error packets or Align error packets received that are greater than 1522 bytes in size
Rx size64	Total number of packets received, including bad packets, that are between 0 and 64 bytes in size
Rx size65to127	Total number of packets received, including bad packets, that are between 65 and 127 bytes in size
Rx size128to255	Total number of packets received, including bad packets, that are between 128 and 255 bytes in size
Rx size256to511	Total number of packets received, including bad packets, that are between 256 and 511 bytes in size
Rx size512to1023	Total number of packets received, including bad packets, that are between 512 and 1023 bytes in size
Rx size1024to1518	Total number of packets received, including bad packets, that are between 1024 and 1518 bytes in size

Table 8-4 Access Area and Network Area Items (continued)

Item	Description
Rx tokenDrop	Total number of packets dropped due to lack of resources (for example, FIFO overflow)
Tx excessDefer	Total number of packets delayed from transmitting due to medium being busy
Tx lateCollision	Number of times that collisions occurred later than 512 bit times after the start of packet transmission
Tx totalGoodPkt	Total number of good packets (multicast, broadcast, and unicast) received by the phone
Tx Collisions	Total number of collisions that occurred while a packet was being transmitted
Tx excessLength	Total number of packets not transmitted because the packet experienced 16 transmission attempts
Tx broadcast	Total number of broadcast packets transmitted by the phone
Tx multicast	Total number of multicast packets transmitted by the phone
LLDP FramesOutTotal	Total number of LLDP frames sent out from the phone
LLDP AgeoutsTotal	Total number of LLDP frames that have been time out in cache
LLDP FramesDiscardedTotal	Total number of LLDP frames that are discarded when any of the mandatory TLVs is missing or out of order or contains out of range string length.
LLDP FramesInErrorsTotal	Total number of LLDP frames that received with one or more detectable errors
LLDP FramesOutTotal	Total number of LLDP frames sent out from the phone
LLDP AgeoutsTotal	Total number of LLDP frames that have been time out in cache
LLDP FramesDiscardedTotal	Total number of LLDP frames that are discarded when any of the mandatory TLVs is missing or out of order or contains out of range string length.
LLDP FramesInErrorsTotal	Total number of LLDP frames that received with one or more detectable errors
LLDP FramesInTotal	Total number of LLDP frames received on the phone.
LLDP TLVDiscardedTotal	Total number of LLDP TLVs that are discarded.
LLDP TLVUnrecognizedTotal	Total number of LLDP TLVs that are not recognized on the phone.
CDP Neighbor Device ID	Identifier of a device connected to this port discovered by CDP protocol.
CDP Neighbor IP Address	IP address of the neighbor device discovered by CDP protocol.
CDP Neighbor Port	Neighbor device port to which the phone is connected discovered by CDP protocol.
LLDP Neighbor Device ID	Identifier of a device connected to this port discovered by LLDP protocol.
LLDP Neighbor IP Address	IP address of the neighbor device discovered by LLDP protocol.
LLDP Neighbor Port	Neighbor device port to which the phone is connected discovered by LLDP protocol.

Device Logs

The Device Logs area on a phone's web page provides information you can use to help monitor and troubleshoot the phone.

- Console Logs—Includes hyperlinks to individual log files. The console log files include debug and error messages received on the phone.
- Core Dumps—Includes hyperlinks to individual dump files.
- Status Messages area—Displays up to the 10 most recent status messages that the phone has generated since it was last powered up. You can also see this information from the Status Messages screen on the phone. Table 7-3 on page 7-4 describes the status messages that can appear.

  To display the Status Messages, access the web page for the phone as described in the Accessing the Web Page for a Phone, page 8-1, and then click the **Status Messages** hyperlink.
- Debug Display area—Displays debug messages that might be useful to Cisco TAC if you require assistance with troubleshooting.

Streaming Statistics

A Cisco Unified IP Phone can stream information to and from up to three devices simultaneously. A phone streams information when it is on a a call or running a service that sends or receives audio or data.

The streaming statistics areas on a phone web page provide information about the streams. Most calls use only one stream (Stream 1), but some calls use two or three stream. For example, a barged call uses Stream 1 and Stream 2.

To display a Streaming Statistics area, access the web page for the phone as described in the Accessing the Web Page for a Phone, page 8-1, and then click the **Stream 1**, the **Stream 2**, or the **Stream 3** hyperlink.

Table 8-5 describes the items in the Streaming Statistics areas.

Table 8-5 Streaming Statistics Area Items

Item	Description
Remote Address	IP address and UDP port of the destination of the stream.
Local Address	IP address and UPD port of the phone.
Start Time	Internal time stamp indicating when Cisco Unified Communications Manager requested that the phone start transmitting packets.
Stream Status	Indication of whether streaming is active.
Host Name	Unique, fixed name that is automatically assigned to the phone based on its MAC address.
Sender Packets	Total number of RTP data packets transmitted by the phone since starting this connection. The value is 0 if the connection is set to receive only mode.
Sender Octets	Total number of payload octets transmitted in RTP data packets by the phone since starting this connection. The value is 0 if the connection is set to receive only mode.
Sender Codec	Type of audio encoding used for the transmitted stream.
Sender Reports Sent[1]	Number of times the RTCP Sender Reports have been sent.

Table 8-5 Streaming Statistics Area Items (continued)

Item	Description
Sender Report Time Sent[1]	Internal time stamp indication when a RTCP Sender Report was sent.
Rcvr Lost Packets	Total number of RTP data packets that have been lost since starting receiving data on this connection. Defined as the number of expected packets less the number of packets actually received, where the number of received packets includes any that are late or duplicate. The value displays as 0 if the connection was set to send-only mode.
Avg Jitter	Estimate of mean deviation of the RTP data packet inter-arrival time, measured in milliseconds. The value displays as 0 if the connection was set to send-only mode.
Rcvr Codec	Type of audio encoding used for the received stream.
Rcvr Reports Sent[1]	Number of times the RTCP Receiver Reports have been sent.
Rcvr Report Time Sent[1]	Internal time stamp indication when a RTCP Receiver Report was sent.
Rcvr Packets	Total number of RTP data packets received by the phone since starting receiving data on this connection. Includes packets received from different sources if this is a multicast call. The value displays as 0 if the connection was set to send-only mode.
Rcvr Octets	Total number of payload octets received in RTP data packets by the device since starting reception on the connection. Includes packets received from different sources if this is a multicast call. The value displays as 0 if the connection was set to send-only mode.
MOS LQK	Score that is an objective estimate of the mean opinion score (MOS) for listening quality (LQK) that rates from 5 (excellent) to 1 (bad). This score is based on audible concealment events due to frame loss in the preceding 8-second interval of the voice stream. For more information, see Monitoring the Voice Quality of Calls, page 9-14. **Note** The MOS LQK score can vary based on the type of codec that the Cisco Unified IP Phone uses.
Avg MOS LQK	Average MOS LQK score observed for the entire voice stream.
Min MOS LQK	Lowest MOS LQK score observed from start of the voice stream.
Max MOS LQK	Baseline or highest MOS LQK score observed from start of the voice stream. These codecs provide the following maximum MOS LQK score under normal conditions with no frame loss: • G.711 gives 4.5 • G.729 A /AB gives 3.8 • G.728/iLBC gives 3.9
MOS LQK Version	Version of the Cisco proprietary algorithm used to calculate MOS LQK scores.
Cumulative Conceal Ratio	Total number of concealment frames divided by total number of speech frames received from start of the voice stream.
Interval Conceal Ratio	Ratio of concealment frames to speech frames in preceding 3-second interval of active speech. If using voice activity detection (VAD), a longer interval might be required to accumulate 3 seconds of active speech.
Max Conceal Ratio	Highest interval concealment ratio from start of the voice stream.
Conceal Secs	Number of seconds that have concealment events (lost frames) from the start of the voice stream (includes severely concealed seconds).

Table 8-5 Streaming Statistics Area Items (continued)

Item	Description
Severely Conceal Secs	Number of seconds that have more than 5 percent concealment events (lost frames) from the start of the voice stream.
Latency[1]	Estimate of the network latency, expressed in milliseconds. Represents a running average of the round-trip delay, measured when RTCP receiver report blocks are received.
Max Jitter	Maximum value of instantaneous jitter, in milliseconds.
Sender Size	RTP packet size, in milliseconds, for the transmitted stream.
Sender Reports Received[1]	Number of times RTCP Sender Reports have been received.
Sender Report Time Received[1]	Last time at which an RTCP Sender Report was received.
Rcvr Size	RTP packet size, in milliseconds, for the received stream.
Rcvr Discarded	RTP packets received from network but discarded from jitter buffers.
Rcvr Reports Received[1]	Number of times RTCP Receiver Reports have been received.
Rcvr Report Time Received[1]	Last time at which an RTCP Receiver Report was received.
Voice Quality Metrics	
MOS LQK	Score that is an objective estimate of the mean opinion score (MOS) for listening quality (LQK) that rates from 5 (excellent) to 1 (bad). This score is based on audible concealment events due to frame loss in the preceding 8-second interval of the voice stream. For more information, see Monitoring the Voice Quality of Calls, page 9-14. The MOS LQK score can vary based on the type of codec that the Cisco Unified IP Phone uses.
Avg MOS LQK	Average MOS LQK score observed for the entire voice stream.
Min MOS LQK	Lowest MOS LQK score observed from start of the voice stream.
Max MOS LQK	Baseline or highest MOS LQK score observed from start of the voice stream. These codecs provide the following maximum MOS LQK score under normal conditions with no frame loss: • G.711 gives 4.5 • G.729 A /AB gives 3.7
MOS LQK Version	Version of the Cisco proprietary algorithm used to calculate MOS LQK scores.
Cmltve Conceal Ratio	Total number of concealment frames divided by total number of speech frames received from start of the voice stream.
Interval Conceal Ratio	Ratio of concealment frames to speech frames in preceding 3-second interval of active speech. If using voice activity detection (VAD), a longer interval might be required to accumulate 3 seconds of active speech.
Max Conceal Ratio	Highest interval concealment ratio from start of the voice stream.

Table 8-5 Streaming Statistics Area Items (continued)

Item	Description
Conceal Secs	Number of seconds that have concealment events (lost frames) from the start of the voice stream (includes severely concealed seconds).
Severely Conceal Secs	Number of seconds that have more than 5 percent concealment events (lost frames) from the start of the voice stream.

1. When the RTP Control Protocol is disabled, no data generates for this field, so it displays as 0.

Related Topics

- Configuring Settings on the Cisco Unified IP Phone
- Configuring Features, Templates, Services, and Users
- Call Statistics Screen, page 7-12
- Monitoring the Voice Quality of Calls, page 9-14

CHAPTER 9

Troubleshooting and Maintenance

This chapter provides information that can assist you in troubleshooting problems with your Cisco Unified IP Phone 7906G or 7911G or with your Cisco Unified Communications network. It also explains how to clean and maintain your phone.

For additional troubleshooting information, refer to the *Using the 79xx Status Information For Troubleshooting* tech note. That document is available to registered Cisco.com users at this URL:

http://www.cisco.com/warp/customer/788/AVVID/telecaster_trouble.html

This chapter includes these topics:

- Resolving Startup Problems, page 9-1
- Cisco Unified IP Phone Resets Unexpectedly, page 9-6
- Troubleshooting Cisco Unified IP Phone Security, page 9-8
- General Troubleshooting Tips, page 9-10
- Resetting or Restoring the Cisco Unified IP Phone, page 9-12
- Using the Quality Report Tool, page 9-14
- Monitoring the Voice Quality of Calls, page 9-14
- Where to Go for More Troubleshooting Information, page 9-16
- Cleaning the Cisco Unified IP Phone, page 9-16

Resolving Startup Problems

After installing a Cisco Unified IP Phone into your network and adding it to Cisco Unified Communications Manager, the phone should start up as described in the Verifying the Phone Startup Process, page 3-11. If the phone does not start up properly, see the following sections for troubleshooting information:

- Symptom: The Cisco Unified IP Phone Does Not Go Through its Normal Startup Process, page 9-2
- Symptom: The Cisco Unified IP Phone Does Not Register with Cisco Unified Communications Manager, page 9-2

Symptom: The Cisco Unified IP Phone Does Not Go Through its Normal Startup Process

When you connect a Cisco Unified IP Phone into the network port, the phone should go through its normal startup process and the LCD screen should display information. If the phone does not go through the startup process, the cause may be faulty cables, bad connections, network outages, lack of power, and so on. Or, the phone may not be functional.

To determine whether the phone is functional, follow these suggestions to systematically eliminate these other potential problems:

1. Verify that the network port is functional:
   - Exchange the Ethernet cables with cables that you know are functional.
   - Disconnect a functioning Cisco Unified IP Phone from another port and connect it to this network port to verify the port is active.
   - Connect the Cisco Unified IP Phone that will not start up to a different network port that is known to be good.
   - Connect the Cisco Unified IP Phone that will not start up directly to the port on the switch, eliminating the patch panel connection in the office.
2. Verify that the phone is receiving power:
   - If you are using external power, verify that the electrical outlet is functional.
   - If you are using in-line power, use the external power supply instead.
   - If you are using the external power supply, switch with a unit that you know to be functional.
3. If the phone still does not start up properly, power up the phone with the handset off-hook. When the phone is powered up in this way, it attempts to launch a backup software image.
4. If the phone still does not start up properly, perform a factory reset of the phone. For instructions, see Performing a Factory Reset, page 9-13.

If after attempting these solutions, the LCD screen on the Cisco Unified IP Phone does not display any characters after at least five minutes, contact a Cisco technical support representative for additional assistance.

Symptom: The Cisco Unified IP Phone Does Not Register with Cisco Unified Communications Manager

If the phone proceeds past the first stage of the startup process (LED buttons flashing on and off) but continues to cycle through the messages displaying on the LCD screen, the phone is not starting up properly. The phone cannot successfully start up unless it is connected to the Ethernet network and it has registered with a Cisco Unified Communications Manager server.

These sections can assist you in determining the reason the phone is unable to start up properly:

- Identifying Error Messages, page 9-3
- Registering the Phone with Cisco Unified Communications Manager, page 9-3
- Checking Network Connectivity, page 9-3
- Verifying TFTP Server Settings, page 9-3
- Verifying IP Addressing and Routing, page 9-3

- Verifying DNS Settings, page 9-4
- Verifying Cisco Unified Communications Manager Settings, page 9-4
- Cisco CallManager and TFTP Services Are Not Running, page 9-4
- Creating a New Configuration File, page 9-5

Identifying Error Messages

As the phone cycles through the startup process, you can access status messages that might provide you with information about the cause of a problem. See Status Messages Screen, page 7-3 for instructions about accessing status messages and for a list of potential errors, their explanations, and their solutions.

Registering the Phone with Cisco Unified Communications Manager

A Cisco Unified IP Phone can register with a Cisco Unified Communications Manager server only if the phone has been added to the server or if auto-registration is enabled. Review the information and procedures in the Adding Phones to the Cisco Unified Communications Manager Database, page 2-7 to ensure that the phone has been added to the Cisco Unified Communications Manager database.

To verify that the phone is in the Cisco Unified Communications Manager database, choose **Device > Find** from Cisco Unified Communications Manager Administration to search for the phone based on its MAC Address. For information about determining a MAC address, see Determining the MAC Address of a Cisco Unified IP Phone, page 2-11.

If the phone is already in the Cisco Unified Communications Manager database, its configuration file may be damaged. See Creating a New Configuration File, page 9-5 for assistance.

Checking Network Connectivity

If the network is down between the phone and the TFTP server or Cisco Unified Communications Manager, the phone cannot start up properly. Ensure that the network is currently running.

Verifying TFTP Server Settings

You can determine the IP address of the TFTP server used by the phone by pressing the **Settings** button on the phone, choosing **IPv4 > Network Configuration**, and scrolling to the **TFTP Server 1** option.

If you have assigned a static IP address to the phone, you must manually enter a setting for the TFTP Server 1 option. See Network Configuration Menu, page 4-5.

If you are using DHCP, the phone obtains the address for the TFTP server from the DHCP server. Check the IP address configured in Option 150.

You can also enable the phone to use an alternate TFTP server. Such a setting is particularly useful if the phone was recently moved from one location to another. See Network Configuration Menu, page 4-5 for instructions.

Verifying IP Addressing and Routing

You should verify the IP addressing and routing settings on the phone. If you are using DHCP, the DHCP server should provide these values. If you have assigned a static IP address to the phone, you must enter these values manually.

On the Cisco Unified IP Phone, press the **Applications Menu** button, then select **Settings > Network Configuration**, and look at the following options:

- DHCP Server—If you have assigned a static IP address to the phone, you do not need to enter a value for the DHCP Server option. However, if you are using a DHCP server, this option must have a value. If it does not, check your IP routing and VLAN configuration. Refer to *Troubleshooting Switch Port Problems*, available at this URL: http://www.cisco.com/warp/customer/473/53.shtml
- IP Address, Subnet Mask, Default Router—If you have assigned a static IP address to the phone, you must manually enter settings for these options. See Network Configuration Menu, page 4-5 for instructions.

If you are using DHCP, check the IP addresses distributed by your DHCP server. Refer to *Understanding and Troubleshooting DHCP in Catalyst Switch or Enterprise Networks*, available at this URL: http://www.cisco.com/warp/customer/473/100.html#41

Verifying DNS Settings

If you are using DNS to refer to the TFTP server or to Cisco Unified Communications Manager, you must ensure that you have specified a DNS server. Verify this setting by pressing the **Applications Menu** button and selecting **Settings > Network Configuration > DNS Server 1**. You should also verify that there is a CNAME entry in the DNS server for the TFTP server and for the Cisco Unified Communications Manager system.

You must also ensure that DNS is configured to do reverse look-ups. Windows2000 is configured by default only to perform forward look-ups.

Verifying Cisco Unified Communications Manager Settings

On the Cisco Unified IP Phone, press the **Applications Menu** button and select **Settings > Network Configuration > Communications Manager 1–5**. The Cisco Unified IP Phone attempts to open a TCP connection to all the Cisco Unified Communications Manager servers that are part of the assigned Cisco Unified Communications Manager group. If none of these options contain IP addresses or show Active or Standby, the phone is not properly registered with Cisco Unified Communications Manager. See Registering the Phone with Cisco Unified Communications Manager, page 9-3 for tips on resolving this problem.

Cisco CallManager and TFTP Services Are Not Running

If the Cisco CallManager or TFTP services are not running, phones may not be able to start up properly. However, in such a situation, it is likely that you are experiencing a system-wide failure, and other phones and devices are unable to start up properly.

If the Cisco CallManager service is not running, all devices on the network that rely on it to make phone calls will be affected. If the TFTP service is not running, many devices will not be able to start up successfully.

Note A service must be activated before it can be started or stopped. To activate a service, choose **Tools > Service Activation**.

<<Reviewers: This procedure is different in the SIP Guide; can you verify which procedure is correct?>>

To start a service, follow these steps:

Procedure

Step 1 From Cisco Unified Communications Manager Administration, choose **Application > Cisco Unified Communications Manager Serviceability**.

Step 2 Choose **Tools > Control Center**.

Step 3 From the Servers column, choose the primary Cisco Unified Communications Manager server.

The page displays the service names for the server that you chose, the status of the services, and a service control panel to stop or start a service.

Step 4 If a service has stopped, click the **Start** button.

The Service Status symbol changes from a square to an arrow.

Creating a New Configuration File

If you continue to have problems with a particular phone that other suggestions in this chapter do not resolve, the configuration file may be corrupted. To create a new configuration file, follow these steps:

Procedure

Step 1 From Cisco Unified Communications Manager, choose **Device > Phone > Find** to locate the phone experiencing problems.

Step 2 Choose **Delete** to remove the phone from the Cisco Unified Communications Manager database.

Step 3 Add the phone back to the Cisco Unified Communications Manager database. See Adding Phones to the Cisco Unified Communications Manager Database, page 2-7 for details.

Step 4 Power cycle the phone.

Note

- When you remove a phone from the Cisco Unified Communications Manager database, its configuration file is deleted from the Cisco Unified Communications Manager TFTP server. The phone's directory number or numbers remain in the Cisco Unified Communications Manager database. They are called "unassigned DNs" and can be used for other devices. If unassigned DNs are not used by other devices, delete them from the Cisco Unified Communications Manager database. You can use the Route Plan Report to view and delete unassigned reference numbers. Refer to *Cisco Unified Communications Manager Administration Guide* for more information.
- Changing the buttons on a phone button template, or assigning a different phone button template to a phone, may result in directory numbers that are no longer accessible from the phone. The directory numbers are still assigned to the phone in the Cisco Unified Communications Manager database, but there is no button on the phone with which calls can be answered. These directory numbers should be removed from the phone and deleted if necessary.

Registering the Phone with Cisco Unified Communications Manager

A Cisco Unified IP Phone can register with a Cisco Unified Communications Manager server only if the phone has been added to the server or if auto-registration is enabled. Review the information and procedures in the Adding Phones to the Cisco Unified Communications Manager Database, page 2-7 to ensure that the phone has been added to the Cisco Unified Communications Manager database.

To verify that the phone is in the Cisco Unified Communications Manager database, choose **Device > Phone > Find** from Cisco Unified Communications Manager Administration to search for the phone based on its MAC Address. For information about determining a MAC address, see Determining the MAC Address of a Cisco Unified IP Phone, page 2-11.

If the phone is already in the Cisco Unified Communications Manager database, its configuration file may be damaged. See Creating a New Configuration File, page 9-5 for assistance.

Cisco Unified IP Phone Resets Unexpectedly

If users report that their phones are resetting during calls or while idle on their desk, you should investigate the cause. If the network connection and Cisco Unified Communications Manager connection are stable, a Cisco Unified IP Phone should not reset on its own.

Typically, a phone resets if it has problems connecting to the Ethernet network or to Cisco Unified Communications Manager. These sections can help you identify the cause of a phone resetting in your network:

- Verifying Physical Connection, page 9-6
- Identifying Intermittent Network Outages, page 9-6
- Verifying DHCP Settings, page 9-7
- Checking Static IP Address Settings, page 9-7
- Verifying Voice VLAN Configuration, page 9-7
- Verifying that the Phones Have Not Been Intentionally Reset, page 9-7
- Eliminating DNS or Other Connectivity Errors, page 9-8
- Checking Power Connection on SIP Phones, page 9-8

Verifying Physical Connection

Verify that the Ethernet connection to which the Cisco Unified IP Phone is connected is up. For example, check whether the particular port or switch to which the phone is connected is down.

Identifying Intermittent Network Outages

Intermittent network outages affect data and voice traffic differently. Your network might have been experiencing intermittent outages without detection. If so, data traffic can resend lost packets and verify that packets are received and transmitted. However, voice traffic cannot recapture lost packets. Rather than retransmitting a lost network connection, the phone resets and attempts to reconnect its network connection.

If you are experiencing problems with the voice network, you should investigate whether an existing problem is simply being exposed.

Verifying DHCP Settings

The following suggestions can help you determine if the phone has been properly configured to use DHCP:

1. Verify that you have properly configured the phone to use DHCP. See Network Configuration Menu, page 4-5 for more information.
2. Verify that the DHCP server has been set up properly.
3. Verify the DHCP lease duration. Cisco recommends that you set it to 8 days.

   Cisco Unified IP Phones send messages with request type 151 to renew their DHCP address leases. If the DHCP server expects messages with request type 150, the lease will be denied, forcing the phone to restart and request a new IP address from the DHCP server.

Checking Static IP Address Settings

If the phone has been assigned a static IP address, verify that you have entered the correct settings. See Network Configuration Menu, page 4-5 for more information.

Verifying Voice VLAN Configuration

If the Cisco Unified IP Phone appears to reset during heavy network usage (for example, following extensive web surfing on a computer connected to same switch as phone), it is likely that you do not have a voice VLAN configured. Isolating the phones on a separate auxiliary VLAN increases the quality of the voice traffic.

Verifying that the Phones Have Not Been Intentionally Reset

If you are not the only administrator with access to Cisco Unified Communications Manager, you should verify that no one else has intentionally reset the phones.

You can check whether a Cisco Unified IP Phone received a command from Cisco Unified Communications Manager to reset by pressing the **Applications Menu** button on the phone and choosing **Settings > Status > Network Statistics**. If the phone was recently reset one of these messages appears:

- Reset-Reset—Phone closed due to receiving a Reset/Reset from Cisco Unified Communications Manager administration.
- Reset-Restart—Phone closed due to receiving a Reset/Restart from Cisco Unified Communications Manager administration.

Eliminating DNS or Other Connectivity Errors

If the phone continues to reset, follow these steps to eliminate DNS or other connectivity errors:

Procedure

Step 1 Use the **Erase** softkey to reset phone settings to their default values. See Resetting or Restoring the Cisco Unified IP Phone, page 9-12 for details.

Step 2 Modify DHCP and IP settings.

a. Disable DHCP. See Network Configuration Menu, page 4-5 for instructions.

b. Assign static IP values to the phone. See Network Configuration Menu, page 4-5 for instructions. Use the same default router setting used for other functioning Cisco Unified IP Phones.

c. Assign TFTP server. See Network Configuration Menu, page 4-5 for instructions. Use the same TFTP server used for other functioning Cisco Unified IP Phones.

Step 3 On the Cisco Unified Communications Manager server, verify that the local host files have the correct Cisco Unified Communications Manager server name mapped to the correct IP address. Refer to *Configuring The IP Hosts File on a Windows 2000 Communications Manager Server*, available at this URL: http://www.cisco.com/warp/customer/788/AVVID/cm_hosts_file.html

Step 4 From Cisco Unified Communications Manager, choose **System > Server** and verify that the server is referred to by its IP address and not by its DNS name.

Step 5 From Cisco Unified Communications Manager, choose **Device > Phone** and verify that you have assigned the correct MAC address to this Cisco Unified IP Phone. For information about determining a MAC address, see Determining the MAC Address of a Cisco Unified IP Phone, page 2-11.

Step 6 Power cycle the phone.

Checking Power Connection on SIP Phones

In most cases, a phone will restart if it powers up using external power but loses that connection and switches to PoE. Similarly, a phone may restart if it powers up using PoE and then gets connected to an external power supply.

Troubleshooting Cisco Unified IP Phone Security

Table 9-1 provides troubleshooting information for the security features on the Cisco Unified IP Phone. For information relating to the solutions for any of these issues, and for additional troubleshooting information about security, refer to *Cisco Unified Communications Manager Security Guide*.

Table 9-1 Cisco Unified IP Phone Security Troubleshooting

Problem	Possible Cause
Device authentication error.	CTL file does not have a Cisco Unified Communications Manager certificate or has an incorrect certificate.
Phone cannot authenticate CTL file.	The security token that signed the updated CTL file does not exist in the CTL file on the phone.
Phone cannot authenticate any of the configuration files other than the CTL file.	Invalid TFTP record.
Phone cannot authenticate any of the configuration file other than ITL file.	The configuration file may not be signed by the corresponding certificate in the phone's Trust List.
Phone does not register with Cisco Unified Communications Manager.	The CTL file does not contain the correct information for the Cisco Unified Communications Manager server.
Phone does not request signed configuration files.	The CTL file does not contain any TFTP entries with certificates.
802.1X Enabled on Phone but Not Authenticating	
Phone cannot obtain a DHCP-assigned IP address.	These errors typically indicate that 802.1X authentication is enabled on the phone, but the phone is unable to authenticate. 1. Verify that you have properly configured the required components (see Supporting 802.1X Authentication on Cisco Unified IP Phones, page 1-16 for more information). 2. Confirm that the shared secret is configured on the phone (see 802.1X Authentication and Status, page 4-37 for more information). – If the shared secret is configured, verify that you have the same shared secret entered on the authentication server. – If the shared secret is not configured, enter it, and ensure that it matches the one on the authentication server.
Phone does not register with Cisco Unified Communications Manager.	
802.1X Authentication Status displays as "Held" (see 802.1X Authentication and Status, page 4-37).	
Status menu displays 802.1X status as "Failed" (see Status Menu, page 7-2).	
802.1X Not Enabled	
Phone cannot obtain a DHCP-assigned IP address.	These errors typically indicate that 802.1X is not enabled on the phone. To enable it, see Security Configuration Menu, page 4-27.
Phone does not register with Cisco Unified Communications Manager.	
Phone status display as "Configuring IP" or "Registering".	
802.1X Authentication Status displays as "Disabled".	
Status menu displays DHCP status as timing out.	
Factory Reset Deleted 802.1X Shared Secret	

Table 9-1 Cisco Unified IP Phone Security Troubleshooting (continued)

Problem	Possible Cause
Phone cannot obtain a DHCP-assigned IP address. Phone does not register with Cisco Unified Communications Manager. Phone status display as "Configuring IP" or "Registering." Cannot access phone menus to verify 802.1X status.	These errors typically indicate that the phone completed a factory reset (see Performing a Factory Reset, page 9-13) while 802.1X was enabled. A factory reset deletes the shared secret, which is required for 802.1X authentication and network access. To resolve this issue, you have two options: • Temporarily disable 802.1X on the switch • Temporarily move the phone to a network environment that is not using 802.1X authentication When the phone starts up normally in one of these conditions, you can access the 802.1X configuration menus and re-enter the shared secret (see 802.1X Authentication and Status, page 4-37).

General Troubleshooting Tips

This section provides troubleshooting information for some common issues that might occur on the Cisco Unified IP Phone.

Table 9-2 provides general troubleshooting information for the Cisco Unified IP Phone.

Table 9-2 Cisco Unified IP Phone Troubleshooting

Summary	Explanation
Daisy-chaining IP phones.	Cisco does not support connecting an IP phone to another IP phone through the PC port. Each IP phone should directly connect to a switch port. If phones are connected together in a line (daisy chaining by using the PC port), the phones will not work.
Poor quality when calling digital cell phones using the G.729 protocol.	In Cisco Unified Communications Manager, you can configure the network to use the G.729 protocol (the default is G.711). When using G.729, calls between an IP phone and a digital cellular phone will have poor voice quality. Use G.729 only when absolutely necessary.
Prolonged broadcast storms cause IP phones to re-register.	Prolonged broadcast storms (lasting several minutes) on the voice VLAN cause the IP phones to re-register with another Cisco Unified Communications Manager server.
Moving a network connection from the phone to a workstation.	If you are powering your phone through the network connection, you must be careful if you decide to unplug the phone's network connection and plug the cable into a desktop computer. **Caution** The computer's network card cannot receive power through the network connection; if power comes through the connection, the network card can be destroyed. To protect a network card, wait 10 seconds or longer after unplugging the cable from the phone before plugging it into a computer. This delay gives the switch enough time to recognize that there is no longer a phone on the line and to stop providing power to the cable.
Changing the telephone configuration.	By default, the network configuration options are locked to prevent users from making changes that could impact their network connectivity. You must unlock the network configuration options before you can configure them. See Unlocking and Locking Options, page 4-3 for details.

Table 9-2 Cisco Unified IP Phone Troubleshooting (continued)

Summary	Explanation
Phone resetting.	The phone resets when it loses contact with the Cisco Unified Communications Manager software. This lost connection can be due to any network connectivity disruption, including cable breaks, switch outages, and switch reboots.
LCD display issues.	If the display appears to have rolling lines or a wavy pattern, it might be interacting with certain types of older fluorescent lights in the building. Moving the phone away from the lights, or replacing the lights, should resolve the problem.
Dual-Tone Multi-Frequency (DTMF) delay.	When you are on a call that requires keypad input, if you press the keys too quickly, some of them might not be recognized.
Loopback condition.	A loopback condition can occur when the following conditions are met: • The SW Port Configuration option in the Network Configuration menu on the phone is set to **10 Half** (10-BaseT / half duplex) • The phone receives power from an external power supply. • The phone is powered down (the power supply is disconnected). In this case, the switch port on the phone can become disabled and the following message will appear in the switch console log: HALF_DUX_COLLISION_EXCEED_THRESHOLD To resolve this problem, re-enable the port from the switch.
Peer to peer image distribution fails.	If the peer to peer image distribution fails, the phone will default to using the TFTP server to download firmware. Access the log messages stored on the remote logging machine to help debug the peer to peer image distribution feature. **Note** These log messages are different than the log messages sent to the phone log.
Cisco VT Advantage/Unified Video Advantage (CVTA)	If you are having problems getting CVTA to work, make sure that the PC Port is enabled, and that CDP is enabled on the PC port. See Network Configuration Menu, page 4-5. (applies to 7911G only)

Table 9-2 Cisco Unified IP Phone Troubleshooting (continued)

Summary	Explanation
Phone call cannot be established	The phone does not have a DHCP IP address, is unable to register to Cisco Unified Communications Manager, and shows a Configuring IP or Registering message. Verify the following: 1. The Ethernet cable is attached. 2. The CCM service is running on the Cisco Unified Communications Manager server. 3. Both phones are registered to the same Cisco Unified Communications Manager. 4. Audio server debug and capture logs are enabled for both phones. If needed, enable Java debug. <<REVIEWERS: Is other action recommended after these items are checked?>>
Call established with the iLBC protocol does not show that the iLBC codec is being used	Call statistics display does not show iLBC as the receiver/sender codec. 1. Check the following using Cisco Unified Communications Manager Administration: – Both phones are in the iLBC device pool. – The iLBC device pool is configured with the iLBC region. – The iLBC region is configured with the iLBC codec. 2. Capture a sniffer trace between the phone and Cisco Unified Communications Manager and verify that SCCP messages,OpenReceiveChannel, and StationMediaTransmit messages have media payload type value equal to 86. If so, the problem is with the phone; otherwise, the problem is with the Cisco Unified Communications Manager configuration. 3. Enable audio server debug and capture logs from both phones. If needed, enable Java debug. <<REVIEWERS: Is other action recommended after these items are checked?>>

Resetting or Restoring the Cisco Unified IP Phone

There are two methods for resetting or restoring the Cisco Unified IP Phone:

- Performing a Basic Reset, page 9-12
- Performing a Factory Reset, page 9-13

Performing a Basic Reset

Performing a basic reset of a Cisco Unified IP Phone provides a way to recover if the phone experiences an error and provides a way to reset or restore various configuration and security settings.

Table 9-3 describes the ways to perform a basic reset. You can reset a phone with any of these operations any time after the phone has started up. Choose the operation that is appropriate for your situation.

Table 9-3 Basic Reset Methods

Operation	Performing	Explanation
Reset phone	From any screen (but not when the phone is idle), press **#**.	Resets any user and network configuration changes that you have made but that the phone has not written to its Flash memory to previously saved settings, then restarts the phone.
Erase softkey	From the Settings menu, unlock phone options (see Unlocking and Locking Options, page 4-3). Then press the **Erase** softkey.	Resets user and network configuration settings to their default values, deletes the CTL file from the phone, and restarts the phone.
	From the Network Configuration menu, unlock phone options (see Unlocking and Locking Options, page 4-3). The press the **Erase** softkey.	Resets network configuration settings to their default values and resets the phone. (This method causes DHCP reconfigure the IP address of the phone.)
	From the Security Configuration menu, unlock phone options (see Unlocking and Locking Options, page 4-3). Then press the **Erase** softkey.	Deletes the CTL file from the phone and restarts the phone.

Performing a Factory Reset

When you perform a factory reset of the Cisco Unified IP Phone, the following information is erased or reset to its default value:

- CTL file—Erased
- User configuration settings—Reset to default values
- Network configuration settings—Reset to default values
- Call histories—Erased
- Locale information—Reset to default values
- Phone application—Erased (phone recovers by loading the term11.default.loads file)

Note This phone must be on a DHCP-enabled network before you can perform these steps.

To perform a factory reset of a phone, follow these steps:

Procedure

Step 1 Unplug the power cable from the phone and then plug it back in.

The phone begins its power-up cycle.

Step 2 While the phone is powering up, and before the **Applications Menu** button flashes on and off, press and hold **#**.

Continue to hold # until the message LED on the handset flashes on and off in sequence in red.

Step 3 Release **#** and press **123456789*0#**.

You can press a key twice in a row, but if you press the keys out of sequence, the factory reset will not take place.

After you press these keys, the message LED on the handset flashes faster in red, and the phone goes through the factory reset process.

Do not power down the phone until it completes the factory reset process, and the main screen appears.

Using the Quality Report Tool

The Quality Report Tool (QRT) is a voice quality and general problem-reporting tool for the Cisco Unified IP Phone. The QRT feature is installed as part of the Cisco Unified Communications Manager installation.

You can configure Cisco Unified IP Phones with QRT. When you do so, users can report problems with phone calls by pressing the **QRT** softkey. This softkey is available only when the Cisco Unified IP Phone is in the Connected, Connected Conference, Connected Transfer, and/or OnHook states.

When a user presses the **QRT** softkey, a list of problem categories appears. The user selects the appropriate problem category, and this feedback is logged in an XML file. Actual information logged depends on the user selection, and whether the destination device is a Cisco Unified IP Phone.

For more information about using QRT, refer to *Cisco Unified Communications Manager Serviceability Administration Guide* and *Cisco Unified Communications Manager Serviceability System Guide*.

Monitoring the Voice Quality of Calls

To measure the voice quality of calls that are sent and received within the network, Cisco Unified IP Phones use these statistical metrics that are based on concealment events. The DSP plays concealment frames to mask frame loss in the voice packet stream.

- Concealment Ratio metrics—Show the ratio of concealment frames over total speech frames. An interval conceal ratio is calculated every 3 seconds.
- Concealed Second metrics—Show the number of seconds in which the DSP plays concealment frames due to lost frames. A severely “concealed second” is a second in which the DSP plays more than five percent concealment frames.
- MOS-LQK metrics—Use a numeric score to estimate the relative voice listening quality. The Cisco Unified IP Phone calculates the mean opinion score (MOS) for listening quality (LQK) based audible concealment events due to frame loss in the preceding 8 seconds, and includes perceptual weighting factors such as codec type and frame size.

  MOS LQK scores are produced by a Cisco proprietary algorithm, Cisco Voice Transmission Quality (CVTQ) index. Depending on the MOS LQK version number, these scores might be compliant with the International Telecommunications Union (ITU) standard P.564. This standard defines evaluation methods and performance accuracy targets that predict listening quality scores based on observation of actual network impairment.

Note Concealment ratio and concealment seconds are primary measurements based on frame loss while MOS LQK scores project a “human-weighted” version of the same information on a scale from 5 (excellent) to 1 (bad) for measuring listening quality.

Listening quality scores (MOS LQK) relate to the clarity or sound of the received voice signal. Conversational quality scores (MOS CQ such as G.107) include impairment factors, such as delay, that degrade the natural flow of conversation.

You can access voice quality metrics from the Cisco Unified IP Phone by using the Call Statistics screen (see Call Statistics Screen, page 7-12) or remotely by using Streaming Statistics (see Monitoring the Cisco Unified IP Phone Remotely, page 8-1.)

Using Voice Quality Metrics

To use the metrics for monitoring voice quality, note the typical scores under normal conditions of zero packet loss and use the metrics as a baseline for comparison.

It is important to distinguish significant changes from random changes in metrics. Significant changes are scores that change about 0.2 MOS or greater and persist in calls that last longer than 30 seconds. Conceal Ratio changes should indicate greater than 3 percent frame loss.

MOS LQK scores can vary based on the codec that the Cisco Unified IP Phone uses. The following codecs provide these maximum MOS LQK scores under normal conditions with zero frame loss:

- G.711 codec gives 4.5 score
- G.729A/ AB gives 3.8 score
- G.728/iLBC gives 3.9 score

Note

- CVTQ does not support wideband (7 kHz) speech codecs, as ITU has not defined the extension of the technique to wideband. Therefore, MOS scores that correspond to G.711 performance are reported for G.722 calls to allow basic quality monitoring, rather than not reporting an MOS score.
- Reporting G.711-scale MOS scores for wideband calls through the use of CVTQ allows basic quality classifications to be indicated as good/normal or bad/abnormal. Calls with high scores (approximately 4.5) indicate high quality/low packet loss, and lower scores (approximately 3.5) indicate low quality/high packet loss.
- Unlike MOS, the Conceal Ratio and Concealed Seconds metrics remain valid and useful for both wideband and narrowband calls.

A Conceal Ratio of zero indicates that the IP network is delivering frames and packets on time with no loss.

Troubleshooting Tips

When you observe significant and persistent changes to metrics, use Table 9-4 for general troubleshooting information:

Table 9-4 Changes to Voice Quality Metrics

Metric Change	Condition
MOS LQK scores decrease significantly	Network impairment from packet loss or high jitter: • Average MOS LQK decreases could indicate widespread and uniform impairment. • Individual MOS LQK decreases indicate bursty impairment. Cross-check with Conceal Ratio and Conceal Seconds for evidence of packet loss and jitter.
MOS LQK scores decrease significantly	• Check to see if the phone is using a different codec than expected (RxType and TxType). • Check to see if the MOS LQK version changed after a firmware upgrade.
Conceal Ratio and Conceal Seconds increase significantly	• Network impairment from packet loss or high jitter.
Conceal Ratio is near or at zero, but the voice quality is poor.	• Noise or distortion in the audio channel such as echo or audio levels. • Tandem calls that undergo multiple encode/decode such as calls to a cellular network or calling card network. • Acoustic problems coming from a speakerphone, handsfree cellular phone or wireless headset. Check packet transmit (TxCnt) and packet receive (RxCnt) counters to verify that voice packets are flowing.

Note Voice quality metrics do not account for noise or distortion, only frame loss.

Where to Go for More Troubleshooting Information

If you have additional questions about troubleshooting the Cisco Unified IP Phones, these Cisco.com web sites can provide you with more tips.

- Cisco Unified IP Phone Troubleshooting Resources:

  http://www.cisco.com/en/US/products/hw/phones/ps379/tsd_products_support_series_home.html
- Cisco Products and Services (Technical Support and Documentation):

  http://www.cisco.com/en/US/products/sw/voicesw/tsd_products_support_category_home.html

Cleaning the Cisco Unified IP Phone

To clean your Cisco Unified IP phone, use a soft, dry cloth to wipe the phone screen. Do not apply liquids or powders directly on the phone. As with all non-weather-proof electronics, liquids and powders can damage the components and cause failures.

APPENDIX A

Providing Information to Users

If you are a system administrator, you are likely the primary source of information for Cisco Unified IP Phone users in your network or company. It is important to provide current and thorough information to end users.

Cisco recommends that you create a web page on your internal support site that provides end users with important information about their Cisco Unified IP Phones.

Consider including the following types of information on this site:

- How Users Obtain Support for the Cisco Unified IP Phone, page A-1
- How Users Get Copies of Cisco Unified IP Phone Manuals, page A-2
- Accessing Cisco 7900 Series Unified IP Phone eLearning Tutorials for SCCP Phones, page A-2
- How Users Subscribe to Services and Configure Phone Features, page A-2
- How Users Access a Voice Messaging System, page A-3
- How Users Configure Personal Directory Entries, page A-3

How Users Obtain Support for the Cisco Unified IP Phone

To successfully use some of the features on the Cisco Unified IP Phone (including speed dial, services, and voice-messaging system options), users must receive information from you or from your network team or be able to contact you for assistance. Make sure to provide end users with the names of people to contact for assistance and with instructions for contacting those people.

Giving Users Access to the User Options Web Pages

Before a user can access the User Options web pages, you must use Cisco Unified Communications Manager Administration to add the user to a standard Cisco Unified Communications Manager end user group: choose **User Management > User Groups**. For additional information, refer to:

- *Cisco Unified Communications Manager Administration Guide*, User Group Configuration.
- *Cisco Unified Communications Manager System Guide*, Roles and User Groups.

How Users Get Copies of Cisco Unified IP Phone Manuals

You should provide end users with access to user documentation for the Cisco Unified IP Phones. The *Cisco Unified IP Phone 7911G Guide* include detailed user instructions for key phone features.

There are several Cisco Unified IP Phone models available, so to assist users in finding the appropriate documentation on the Cisco website, Cisco recommends that you provide links to the current documentation. If you do not want to or cannot send users to the Cisco website, Cisco suggests that you download the PDF files and provide them to end users on your website.

For a list of available documentation, go to the Cisco Unified IP Phone website at this URL:

http://www.cisco.com/en/US/products/hw/phones/ps379/tsd_products_support_series_home.html

For more information about viewing or ordering documentation, see Document Conventions, page -3.

Accessing Cisco 7900 Series Unified IP Phone eLearning Tutorials for SCCP Phones

Cisco 7900 Series Unified IP Phone eLearning tutorials use audio and animation to demonstrate basic calling features for SCCP phones. The eLearning tutorials are currently available for the Cisco Unified IP Phone 7970 Series (7970G/7971G-GE) and the Cisco Unified IP Phone models 7961G/G-GE, 7941G/G-GE, 7960G, 7940G, 7912G, and 7905G.

End-users can access runtime versions of the eLearning tutorials (English only) from Cisco.com by looking for tutorials under relevant phone models at this site:

http://cisco.com/en/US/products/hw/phones/ps379/products_user_guide_list.html

Administrators can download customizable versions of the eLearning tutorials (English only) from the phone product pages on cisco.com at

http://cisco.com/en/US/products/hw/phones/ps379/prod_models_home.html

Refer to the tutorial Read Me file that is included with the relevant eLearning tutorial for specific instructions, including how to link to the most recent user guide PDF.

Note The eLearning tutorials are updated periodically and therefore might not contain the latest feature information for end-users. For the latest feature information, end-users should refer to the Cisco Unified IP Phone end-user documentation specific to their phone model and Cisco Unified Communications Manager version.

How Users Subscribe to Services and Configure Phone Features

End users can perform a variety of activities using the Cisco Unified Communications Manager User Options web pages. These activities include subscribing to services, setting up speed dial and call forwarding numbers, configuring ring settings, and creating a personal address book. Keep in mind that configuring settings on a phone using a website might be new for your end users. You need to provide as much information as possible to ensure that they can successfully access and use the User Options web pages.

Make sure to provide end users with the following information about the User Options web pages:

- The URL required to access the application. This URL is:

  http://*server_name*/CCMUser/, where *server_name* is the host on which the web server is installed.

- A user ID and default password needed to access the application.

  These settings correspond to the values you entered when you added the user to Cisco Unified Communications Manager (see Adding Users to Cisco Unified Communications Manager, page 5-21).

- A brief description of what a web-based, graphical user interface application is, and how to access it with a web browser.

- An overview of the tasks that users can accomplish using the web page.

You can also refer users to *Customizing Your Cisco Unified IP Phone on the Web*, which is available at this URL:

http://www.cisco.com/en/US/products/hw/phones/ps379/tsd_products_support_series_home.html

How Users Access a Voice Messaging System

Cisco Unified Communications Manager lets you integrate with many different voice mail messaging systems, including the Cisco Unity voice messaging system. Because you can integrate with a variety of systems, you must provide users with information about how to use your specific system.

You should provide this information to each user:

- How to access the voice mail messaging system account.

  Make sure that you have used Cisco Unified Communications Manager to configure the **Messages** menu or the **Msgs** softkey.

- Initial password for accessing the voice messaging system.

  Make sure that you have configured a default voice messaging system password for all users.

- How the phone indicates that voice messages are waiting.

  Make sure that you have used Cisco Unified Communications Manager to set up a message waiting indicator (MWI) method.

How Users Configure Personal Directory Entries

Users can configure personal directory entries on the Cisco Unified IP Phone. To configure a personal directory, users must have access to the following:

- User Options web pages—Make sure that users know how to access their User Options web pages. See How Users Subscribe to Services and Configure Phone Features, page A-2 for details.

- Cisco Unified IP Phone Address Book Synchronizer—Make sure to provide users with the installer for this application. To obtain the installer, choose **Application > Plugins** from Cisco Unified Communications Manager Administration and click **Download**, which is located next to the **Cisco Unified IP Phone Address Book Synchronizer** plugin name. When the file download dialog box displays, click **Save**. Send the TabSyncInstall.exe file to all users who require this application.

See Installing and Configuring the Cisco Unified IP Phone Address Book Synchronizer, page A-4 for information about installing the Cisco Unified IP Phone Address Book Synchronizer.

Installing and Configuring the Cisco Unified IP Phone Address Book Synchronizer

Use this tool to synchronize data stored in your Microsoft Windows address book with the Cisco Unified Communications Manager directory and the User Options Personal Address Book.

Tip To successfully synchronize the Windows address book with the Personal Address Book, all Windows address book users should be entered in the Windows address book before performing the following procedures.

Installing the Synchronizer

Step 1 Get the Cisco Unified IP Phone Address Book Synchronizer installer file from your system administrator.

Step 2 Double-click the TabSyncInstall.exe file provided by your system administrator.

The publisher dialog box displays.

Step 3 Click **Run**.

The Welcome to the InstallShield Wizard for Cisco Unified CallManager Personal Address Book Synchronizer window displays.

Step 4 Click **Next**.

The License Agreement window displays.

Step 5 Read the license agreement information, and click the I Accept radio button. Click **Next**.

The Destination Location window displays.

Step 6 Choose the directory in which you want to install the application and click **Next**.

The Ready to Install window displays.

Step 7 Click **Install**.

The installation wizard installs the application to your computer. When the installation is complete, the InstallShield Wizard Complete window displays.

Step 8 Click **Finish**.

Step 9 To complete the process, follow the steps in the Configuring the Synchronizer, page A-4.

Configuring the Synchronizer

Step 1 Open the Cisco Unified IP Phone Address Book Synchronizer.

If you accepted the default installation directory, you can open the application by choosing **Start > All Programs > Cisco Systems > TabSync**.

Step 2 To configure user information, click the **User** button.

The Cisco Unified CallManager User Information window displays.

Step 3 Enter the Cisco Unified IP Phone user name and password and click **OK**.

Step 4 To configure Cisco Unified Communications Manager server information, click the **Server** button.

The Configure Cisco Unified CallManager Server Information window displays.

Step 5 Enter the IP address or host name and the port number of the Cisco Unified Communications Manager server and click **OK**.

If you do not have this information, contact your system administrator.

Step 6 To start the directory synchronization process, click the **Synchronize** button.

The Synchronization Status window provides information on the status of the address book synchronization. If you chose the user intervention for duplicate entries rule and you have duplicate address book entries, the Duplicate Selection window displays. Choose the entry that you want to include in your Personal Address Book and click **OK**.

When synchronization completes, click **Exit** to close the Cisco Unified CallManager Address Book Synchronizer. To verify if the synchronization worked, log in to your User Options web pages and choose Personal Address Book. The users from your Windows address book should be listed.

APPENDIX B

Feature Support by Protocol for Cisco Unified IP Phone 7906G and 7911G

This appendix provides information about feature support for the Cisco Unified IP Phone 7906G and 7911G using the SCCP or SIP protocol with Cisco Unified Communications Manager Release 7.0.

In most cases, the Cisco Unified IP Phone 7906G and 7911G supports similar features whether on SCCP or SIP. Table B-1 provides a high-level overview of calling features and their support by protocol. This table focuses primarily on end-user calling features and is not intended to represent a comprehensive listing of all available phone features. For details about user interface differences and feature use, refer to *Cisco Unified IP Phone 7906G and 7911G User Guide for Cisco Unified Communications Manager 7.0*, which is available at this URL:

http://www.cisco.com/en/US/products/hw/phones/ps379/tsd_products_support_series_home.html

The specific sections that describe the features in the phone user guide are referenced in Table B-1.

Table B-1 Cisco IP Phone Feature Support by Protocol

Features	Cisco Unified IP Phone 7906G, 7911G		For More Information
	SCCP	SIP	
Calling Features			
Abbreviated Dialing	Supported	Supported	Basic Call Handling—Placing a Call: Additional Options
Anonymous Call Block	Not Supported	Supported	
Audible Message Waiting Indicator	Supported	Supported	Accessing Voice Messages
Auto Answer	Supported	Supported	Using a Handset, Headset, and Speakerphone—Using Auto Answer
Auto Dial	Supported	Supported	Basic Call Handling—Placing a Call: Basic Options
Barge (and cBarge)	Supported	Supported	Advanced Call Handling—Using a Shared Line
Block external-to-external transfers	Supported	Supported	
Call Back	Supported	Supported	Basic Call Handling—Placing a Call: Additional Options
Call Chaperone	Supported	Supported	
Call Display Restrictions	Supported	Supported	

Table B-1 Cisco IP Phone Feature Support by Protocol (continued)

Features	Cisco Unified IP Phone 7906G, 7911G		For More Information
	SCCP	SIP	
Calling Features			
Call Forward All	Supported	Supported	Basic Call Handling—Forwarding Calls to Another Number
Call Forward All Breakout	Supported	Supported	
Call Forward All Loop Prevention	Supported	Supported	
Call Forward Busy	Supported	Supported	Basic Call Handling—Forwarding Calls to Another Number
Call Forward Configurable Display	Supported	Supported	
Call Forward Destination Override	Supported	Supported	
Call Forward No Answer	Supported	Supported	Basic Call Handling—Forwarding Calls to Another Number
Call Park	Supported	Supported	Advanced Call Handling—Storing and Receiving Parked Calls
Call Pickup/Group Call Pickup/Directed Call Pickup/Other Call Pickup	Supported	Supported	Advanced Call Handling—Picking Up a Redirected Call on Your Phone
Call Recording	Supported	Supported	
Call Waiting	Supported	Supported	Basic Call Handling—Answering a Call
Caller ID	Supported	Supported	An Overview of Your Phone—Understanding Touch Screen Features or An Overview of Your Phone—Understanding Phone Screen Features
Caller ID Blocking	Supported	Supported	
Cisco Call Back	Supported	Supported	
Cisco Unified Communications Manager Assistant	Supported	Supported	
Client Matter Codes (CMC)	Supported	Not supported	Basic Call Handling—Placing a Call: Additional Options
Conference	Supported	Supported	Basic Call Handling—Making Conference Calls
Computer Telephony Integration (CTI) Applications	Supported	Some support (such as Call Park, WMI)	Users do not interact with this feature directly. It is configured on Cisco Unified Communications Manager
Do Not Disturb (DND)	Supported	Supported	Basic Call Handling—Using Do Not Disturb
Directed Call Park	Supported	Supported	Advanced Call Handling—Storing and Receiving Parked Calls
Enbloc Dialing	Supported	Not Supported	

Table B-1 Cisco IP Phone Feature Support by Protocol (continued)

Features	Cisco Unified IP Phone 7906G, 7911G		For More Information
	SCCP	SIP	
Calling Features			
Distinctive Ring	Supported	Supported	Using Phone Settings—Customizing Rings and Message Indicators
Extension Mobility	Supported	Supported	Advanced Call Handling—Using Cisco Extension Mobility
Extension Mobility Change PIN	Supported	Supported	Advanced Call Handling—Using Cisco Extension Mobility
Extension Mobility Cross Cluster	Supported	Supported	
External Call Control	Supported	Supported	
Fast Dial Service	Supported	Supported	Advanced Call Handling—Speed Dialing
Forced Authorization Codes (FAC)	Supported	Not supported	Basic Call Handling—Placing a Call: Additional Options
Help System	Supported	Supported	An Overview of Your Phone—Understanding Feature Buttons and Menus
Hold/Resume	Supported	Supported	Basic Call Handling—Using Hold and Resume
Hold Reversion	Supported	Supported	Basic Call Handling—Using Hold and Resume
Hold Status	Supported	Supported	Using Hold and Resume
Hunt Group	Supported	Supported	
Immediate Divert	Supported	Supported	Basic Call Handling—Answering a Call
Immediate Divert—Enhanced	Supported	Supported	Basic Call Handling—Sending a Call to a Voice Messaging System
Intelligent Session Control	Supported	Supported	
Intercom	Supported	Supported	Basic Call Handling—Placing or Receiving Intercom Calls
Inter-Cluster Trust (Bulk Certificate Replication)	Supported	Supported	
Intra-Cluster Trust (Bulk Certificate Replication)	Supported	Supported	
Join/Select	Supported	Not supported	Basic Call Handling—Making Conference Calls
Join Across Lines	Supported	Not supported	Basic Call Handling—Making Conference Calls
Log Out of Hunt Groups	Supported	Supported	Advanced Call Handling—Logging Out of Hunt Groups
Malicious Call ID	Supported	Not supported	Advanced Call Handling—Tracing Suspicious Calls
Meet-Me Conference	Supported	Supported	Basic Call Handling—Making Conference Calls
Message Waiting Indicator	Supported	Supported	
Missed call logging	Supported	Supported	Using Call Logs and Directories
Mobile Connect	Supported	Supported	

Table B-1 Cisco IP Phone Feature Support by Protocol (continued)

Features	Cisco Unified IP Phone 7906G, 7911G		For More Information
	SCCP	SIP	
Calling Features			
Mobile Voice Access	Supported	Supported	
Multilevel Precedence and Preemption (MLPP)	Supported	Not supported	Advanced Call Handling—Prioritizing Critical Calls
Multiple Calls per Line Appearance	200	50	An Overview of Your Phone—Understanding Lines vs. Calls
Mute	Supported	Supported	Basic Call Handling—Using Mute
On-hook Dialing/Pre-Dial	Supported	Supported	Basic Call Handling—Placing a Call: Basic Options
Onhook Call Transfer	Supported	Supported	
Other Group Pickup	Supported	Supported	
Phone secure web access	Supported	Supported	
Presence-Enabled Directories	Supported	Supported	
Private Line Automated Ringdown (PLAR)	Supported	Supported	
Privacy	Supported	Supported	Advanced Call Handling—Using a Shared Line
Programmable Line Keys	Supported	Not supported	Feature descriptions throughout phone guide
Quality Reporting Tool (QRT)	Supported	Supported	Troubleshooting—Using the Quality Reporting Tool
Redial	Supported	Supported	Basic Call Handling—Placing a Call: Basic Options
Ring Setting	Supported	Supported	
Ringer Volume Control	Supported	Supported	
Secure Conference	Supported	Supported	Basic Call Handling—Making Conference Calls
Services	Supported	Supported	
Services URL Button	Supported	Supported	
Session Handoff	Supported	Supported	Basic Call Handling—Switching an In-Progress Call to Another Phone
Shared Line	Supported	Supported	Advanced Call Handling—Using a Shared Line
Silent Monitoring	Supported	Supported	
Single Button Barge	Supported	Not supported	Advanced Call-Handling—Using Barge to Add Yourself to a Shared-Line Call
Speed Dialing	Supported	Supported	Advanced Call Handling—Speed Dialing
Time-of-day Routing	Supported	Supported	
Time Zone Update	Supported	Supported	
Touchscreen Illumination Disabling	Supported	Supported	
Transfer	Supported	Supported	Basic Call Handling—Transferring Calls

Table B-1 Cisco IP Phone Feature Support by Protocol (continued)

Features	Cisco Unified IP Phone 7906G, 7911G		For More Information
	SCCP	SIP	
Calling Features			
URL Dialing	Not supported	Supported	Using Call Logs and Directories—Using Call Logs
Video Support	Supported	Not supported	Understanding Additional Configuration Options
Virtual Private Network Support in Phones	Supported	Supported	
Voice Mail	Supported	Supported	Accessing Voice Messages section of the Phone Guide
WebDialer	Supported	Supported	Customizing Your Phone on the Web—Configuring Features and Services on the Web
Settings			
Call Statistics	Supported	Supported	Troubleshooting Your Phone—Viewing Phone Administrative Data
Voice Quality Metrics	Supported	Supported	Troubleshooting Your Phone—Viewing Phone Administrative Data
Services			
SDK Compliance	4.0(1)	4.0(1)	*Cisco Unified IP Phone Service Application Development Notes for Release 4.1(3)* or later
Directories			
Call Logs	Supported	Supported	Using Call Logs and Directories—Directory Dialing
Corporate Directories	Supported	Supported	Using Call Logs and Directories—Directory Dialing
Personal Directory Enhancements	Supported	Supported	Using Call Logs and Directories—Directory Dialing
Supplemental Features and Applications			
Cisco Unified Communications Manager Assistant	Supported	Supported	*Cisco Unified Communications Manager Assistant User Guide*
Cisco Unified Communications Manager Auto-Attendant	Supported	Not supported	*Cisco Unified Communications Manager Features and Services Guide*
Cisco Unified Business Attendant Console, Cisco Unified Department Attendant Console, or Cisco Unified Enterprise Attendant Console	Supported	Supported	These are third-party products. See http://www.cisco.com/en/US/products/ps7282/prod_maintenance_guides_list.html
Cisco VT Advantage	Supported	Not supported	*Cisco VT Advantage User Guide*

APPENDIX C

Supporting International Users

If you are using Cisco Unified IP Phones in a locale other than English (United States), you must install the locale-specific version of the Cisco Unified Communications Manager Locale Installer on every Cisco Unified Communications Manager server in the cluster. Installing the locale installer ensures that you have the latest translated text, user and network locales, and country-specific phone tones available for the Cisco Unified IP Phones. You can find locale-specific versions of the Cisco Unified Communications Manager Locale Installer at this URL:

http://www.cisco.com/kobayashi/sw-center/telephony/callmgr/locale-installer.shtml.

For more information, refer to the Software Upgrades section in the *Cisco Unified Communications Operating System Administration Guide.*

Note All languages may not be immediately available, so continue to check the website for updates.

Support for International Call Logging

If your phone system is configured for international call logging, the call logs, redial, or call directory entries may display a "+" symbol to represent the international escape code for your location. Depending on the configuration for your phone system, the "+" may be replaced with the correct international dialing code, or you may need to edit the number before dialing to manually replace the "+" with the international escape code for your location. In addition, while the call log or directory entry may display the full international number for the received call, the phone display may show the shortened local version of the number, without international or country codes.

APPENDIX D

Technical Specifications

The following sections describe the technical specifications for the Cisco Unified IP Phones 7906G and 7911G.

- Physical and Operating Environment Specifications, page D-1
- Cable Specifications, page D-2
- Network and Access Port Pinouts, page D-2

Physical and Operating Environment Specifications

Table D-1 shows the physical and operating environment specifications for the Cisco Unified IP Phone.

Table D-1 Physical and Operating Specifications

Specification	Value or Range
Operating temperature	32° to 104°F (0° to 40°C)
Operating relative humidity	10% to 95% (non-condensing)
Storage temperature	14° to 140°F (–10° to 60°C)
Height	6.5 in. (20.3 cm)
Width	7 in. (17.67 cm)
Depth	6 in. (15.2 cm)
Weight	1.9 lb (0.9 kg)
Power options	• The phone can receive power from IEEE 802.3af-compliant data switches (Class III) • The phone can be powered locally with a power adapter (Cisco part number CP-PWR-CUBE-3=) and the appropriate power cord (power requirements for the power adapter: 100-240 VAC, 50-60 Hz, 0.5 A)

Table D-1 Physical and Operating Specifications (continued)

Specification	Value or Range
Cables	Category 3/5/5e for 10-Mbps cables with 4 pairs Category 5/5e for 100-Mbps cables with 4 pairs Category 5e/6 for 1000-Mbps cables with 4 pairs **Note** Cables have 4 pairs of wires for a total of 8 conductors.
Distance requirements	As supported by the Ethernet specification, it is assumed that the maximum cable length between each Cisco Unified IP Phone and the switch is 100 meters (330 feet).

Cable Specifications

- RJ-9 jack (4-conductor) for handset and headset connection.
- RJ-45 jack for the LAN 10/100BaseT connection (labeled 10/100 SW).
- RJ-45 jack for the access port 10/100BaseT connection (labeled 10/100 PC).
- 48-volt power connector.

Network and Access Port Pinouts

Although both the network and access ports are used for network connectivity, they serve different purposes and have different port pinouts.

Network Port Connector

Table D-2 describes the network port connector pinouts.

Table D-2 Network Port Connector Pinouts

Pin Number	Function
1	BI_DA+
2	BI_DA-
3	BI_DB+
4	BI_DC+
5	5BI_DC-
6	6BI_DB-
7	7BI_DD+
8	BI_DD-

Note "BI" stands for bi-directional, while DA, DB, DC and DD stand for "Data A", "Data B", "Data C" and "Data D", respectively.

Access Port Connector

Table D-3 describes the access port connector pinouts.

Table D-3 Access Port Connector Pinouts

Pin Number	Function
1	BI_DB+
2	BI_DB-
3	BI_DA+
4	BI_DD+
5	5BI_DD-
6	6BI_DA-
7	7BI_DC+
8	BI_DC-

Note "BI" stands for bi-directional, while DA, DB, DC and DD stand for "Data A", "Data B", "Data C" and "Data D", respectively.

APPENDIX E

Basic Phone Administration Steps

This appendix provides minimum, basic configuration steps for you to do the following:

- Add a new user to Cisco Unified Communications Manager Administration
- Configure a new phone for that user
- Associate that user to that phone
- Complete other basic end-user configuration tasks

The procedures provide one method for performing these tasks and are not the only way to perform these tasks. They are a streamlined approach to get a new user and corresponding phone running on the system.

These procedures are designed to be used on a mature Cisco Unified Communications Manager system where calling search spaces, partitions, and other complicated configuration have already been done and are in place for existing users.

This section contains these topics:

- Example User Information for these Procedures, page E-1
- Adding a User to Cisco Unified Communications Manager, page E-2
- Configuring the Phone, page E-3
- Performing Final End User Configuration Steps, page E-6

Example User Information for these Procedures

In the procedures that follow, example are given when possible to illustrate some of the steps. Sample user and phone information used throughout these procedures includes:

- User's Name: John Doe
- User ID: johndoe
- Phone model: 7911G
- Protocol: SCCP
- MAC address listed on phone: 00127F576611
- Five-digit internal telephone number: 26640

Adding a User to Cisco Unified Communications Manager

This section describes steps for adding a user to Cisco Unified Communications Manager. Follow one of the procedures in this section, depending on your operating system and the manner in which you are adding the user:

- Adding a User From an External LDAP Directory, page E-2
- Adding a User Directly to Cisco Unified Communications Manager, page E-2

Adding a User From an External LDAP Directory

If you added a user to an LDAP Directory (a non-Cisco Unified Communications Server directory), you can immediately synchronize that directory to the Cisco Unified Communications Manager on which you are adding this same user and the user's phone by following these steps:

Procedure

Step 1 Log onto Cisco Unified Communications Manager Administration.

Step 2 Choose **System > LDAP > LDAP Directory**.

Step 3 Use the **Find** button to locate your LDAP directory.

Step 4 Click on the LDAP directory name.

Step 5 Click **Perform Full Sync Now**.

Note If you do not need to immediately synchronize the LDAP Directory to the Cisco Unified Communications Manager, the LDAP Directory Synchronization Schedule on the LDAP Directory window determines when the next auto-synchronization is scheduled. However, the synchronization must occur before you can associate a new user to a device.

Step 6 Proceed to Configuring the Phone, page E-3

For more information and limitations on configuring LDAP system, refer to the *Cisco Unified Communications Manager Administration Guide*, LDAP System Configuration, LDAP Directory Configuration, and LDAP Authentication Configuration and *Cisco Unified Communications Manager System Guide, Cisco Systems, Inc.* Understanding the Directory.

Adding a User Directly to Cisco Unified Communications Manager

If you are not using an LDAP directory, you can add a user directly to Cisco Unified Communications Manager Administration by following these steps:

Procedure

Step 1 Choose **User Management > End User**, then click **Add New.** The End User Configuration window appears.

Step 2 In the User Information pane of this window, enter the following:

- User ID—Enter the end user identification name. Cisco Unified Communications Manager does not permit modifying the user ID after it is created. You may use the following special characters: =, +, <, >, #, ;, \, , "", and blank spaces.

  Example: *johndoe*

- Password and Confirm Password—Enter five or more alphanumeric or special characters for the end user password. You may use the following special characters: =, +, <, >, #, ;, \, , "", and blank spaces.

- Last Name—Enter the end user last name. You may use the following special characters: =, +, <, >, #, ;, \, , "", and blank spaces.)

  Example: *doe*

- Telephone Number—Enter the primary directory number for the end user. End users can have multiple lines on their phones.

  Example: 26640 (John Doe's internal company telephone number)

Step 3 Click **Save**.

Step 4 Proceed to the section Configuring the Phone, page E-3.

Configuring the Phone

To identify the user's phone model and protocol, follow these steps:

Procedure

Step 1 From Cisco Unified Communications Manager administration, choose **Device > Phone >.**

Step 2 Click **Add New**.

Step 3 Select the user's phone model from the Phone Type drop-down list, then click **Next**.

Step 4 Select the device protocol (SCCP or SIP) from the drop-down list, then click **Next**. The Phone Configuration window appears.

On the Phone Configuration window, you can use the default values for most of the fields.

To configure the required fields and some key additional fields, follow these steps:

Procedure

Step 1 For the required fields, possible values, some of which are based on the example of user *johndoe*, can be configured as follows:

a. In the Device Information pane of this window:

- MAC Address—Enter the MAC address of the phone, which is listed on a sticker on the phone.

  Make sure that the value comprises 12 hexadecimal characters.

  Example: 00127F576611 (MAC address on john doe's phone)

- Description—This is an optional field in which you can enter a useful description, such as *john doe's phone*. This will help you if you need to search on information about this user.
- Device Pool—Choose the device pool to which you want this phone assigned. The device pool defines sets of common characteristics for devices, such as region, date/time group, softkey template, and MLPP information.

Note Device Pools are defined on the Device Pool Configuration window of Cisco Unified Communications Server Administration (**System > Device Pool)**.

- Phone Button Template—Choose the appropriate phone button template from the drop-down list. The phone button template determines the configuration of buttons on a phone and identifies which feature (line, speed dial, and so on) is used for each button.

Note Phone button templates are defined on the Phone Button Template Configuration window of Cisco Unified Communications Manager Administration (**Device > Device Settings > Phone Button Template**). You can use the search field(s) in conjunction with the **Find** button to find all configured phone button templates and their current settings.

- Softkey Template—Choose the appropriate softkey template. The softkey template determines the configuration of the softkeys on Cisco Unified IP Phones. Leave this field blank if the common device configuration contains the assigned softkey template.

Note Softkey templates are defined on the Softkey Template Configuration window of Cisco Unified Communications Manager Administration (**Device > Device Settings > Softkey Template**). You can use the search field(s) in conjunction with the **Find** button to find all configured softkey templates and their current settings.

- Common Phone Profile—From the drop-down list box, choose a common phone profile from the list of available common phone profiles.

Note Common Phone Profiles are defined on the Common Phone Profile Configuration window of Cisco Unified Communications Manager Administration (**Device > Device Settings > Common Phone Profile**). You can use the search field(s) in conjunction with the **Find** button to find all configured common phone profiles and their current settings.

- Calling Search Space—From the drop-down list box, choose the appropriate calling search space (CSS). A calling search space comprises a collection of partitions (analogous to a collection of available phone books) that are searched to determine how a dialed number should be routed. The calling search space for the device and the calling search space for the directory number get used together. The directory number CSS takes precedence over the device CSS.

Note Calling Search Spaces are defined on the Calling Search Space Configuration window of Cisco Unified Communications Manager Administration (**Calling routing > Class of Control> Calling Search Space**). You can use the search field(s) in conjunction with the **Find** button to find all configured Calling Search Spaces and their current settings.

- Location—Choose the appropriate location for this Cisco Unified IP Phone.
- Owner User ID—From the drop-down menu, choose the user ID of the assigned phone user.

b. In the Protocol Specific Information pane of this window, choose a Device Security Profile from the drop-down list. To enable security features for a phone, you must configure a new security profile for the device type and protocol and apply it to the phone. If the phone does not support security, choose a nonsecure profile.

To identify the settings that are contained in the profile, choose **System > Security Profile > Phone Security Profile**.

Note The security profile chosen should be based on the overall security strategy of the company.

c. (For SIP Phones only) Also in the Protocol Specific Information pane of this window, choose the applicable SIP Profile from the drop-down list.

d. In the Extension Information pane of this window, check the Enable Extension Mobility box if this phone supports Cisco Extension Mobility.

e. In the Product Specific Configuration Layout pane of this window, enable the Video Capabilities field if this field appears on your window.

f. Click **Save**.

Step 2 Configure line settings:

a. On the Phone Configuration window, click Line 1 on the left pane of the window. The Directory Number Configuration window appears.

b. In the Directory Number field, enter a valid number that can be dialed.

Note This field should contain the same number that appears in the Telephone Number field on the User Configuration window.

Example: 26640 is the directory number of user John Doe in the example above.

c. From the Route Partition drop-down list, choose the partition to which the directory number belongs. If you do not want to restrict access to the directory number, choose <None> for the partition.

d. From the Calling Search Space drop-down list (Directory Number Settings pane of the Directory Number Configuration window), choose the appropriate calling search space. A calling search space comprises a collection of partitions that are searched for numbers that are called from this directory number. The value that you choose applies to all devices that are using this directory number.

e. In the Call Pickup and Call Forward Settings pane of the Directory Number Configuration window, choose the items (i.e. Forward All, Forward Busy Internal) and corresponding destinations to which calls should be sent.

Example: If you want incoming internal and external calls that receive a busy signal to be forwarded to the voice mail for this line, check the Voice Mail box next to the “Forward Busy Internal” and “Forward Busy External” items in the left column of the Call Pickup and Call Forward Settings pane.

f. In the “Line 1 on Device...” pane of the Directory Number Configuration window, configure the following:

- Display (Internal Caller ID field)—You can enter the first name and last name of the user of this device so that this name will be displayed for all internal calls. You can also leave this field blank to have the system display the phone extension.
- External Phone Number Mask—Indicate phone number (or mask) that is used to send Caller ID information when a call is placed from this line.

  You can enter a maximum of 24 number and "X" characters. The Xs represent the directory number and must appear at the end of the pattern.

  Example: Using the john doe extension in the example above, if you specify a mask of 408902XXXX, an external call from extension 6640 displays a caller ID number of 4089026640.

Note This setting applies only to the current device unless you check the check box at right (Update Shared Device Settings) and click the **Propagate Selected** button. (The check box at right displays only if other devices share this directory number.)

g. Click **Save**.

h. Click **Associate End Users** at the bottom of the window to associate a user to the line being configured. Use the Find button in conjunction with the Search fields to locate the user, then check the box next to the user's name, then click **Add Selected**. The user's name and user ID should now appear in the "Users Associated With Line" pane of the Directory Number Configuration window.

i. Click **Save**. The user is now associated with Line 1 on the phone.

j. If your phone has a second line, configure Line 2.

k. Associate the user with the device:

- Choose **User Management > End User**.
- Use the search boxes and the Find button to locate the user you have added (i.e. *doe* for the last name).
- Click on the user ID (i.e. *johndoe*). The End User Configuration window appears.
- Click **Device Associations**.
- Use the Search fields and the Find button to locate the device with which you want to associate to the user. Select the device, then click **Save Selected/Changes**. The user is now associated with the device.
- Click the **Go** button next to the "Back to User" Related link in the upper-right corner of the screen.

l. Proceed to Performing Final End User Configuration Steps, page E-6.

Performing Final End User Configuration Steps

If you are not already on the End User Configuration page, choose **User Management > End User** to perform some final configuration tasks. Use the Search fields and the Find button to locate the user (i.e. John Doe), then click on the user ID to get to the End User Configuration window for the user.

In the End User configuration window, do the following:

Procedure

Step 1 In the Directory Number Associations pane of the screen, set the primary extension from the drop-down list.

Step 2 In the Mobility Information pane, check the Enable Mobility box.

Step 3 In the Permissions Information pane, use the User Group buttons to add this user to any user groups. For example, you may want to add the user to a group that has been defined as a "Standard CCM End User Group."

To view all configured user groups, choose **User Management > User Group**.

Step 4 Click **Save**.

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